

**User's Manual
For**



&



**DC FORENSICS AUDIO LABORATORY,
Version 8 & DC 8**

User's Manual

For

DIAMOND CUT FORENSICS AUDIO
LABORATORY, Version 8

&

DC 8



DC FORENSICS AUDIO LABORATORY v. 8 / DIAMOND CUT EIGHT

User's Manual

Sixteenth Edition

2nd Printing

DC FORENSICS AUDIO LABORATORY v. 8 & DC 8

DC FORENSICS Audio Laboratory, Version 8.1 and above but less than 9.0

DC8, Version 8.1 and above but less than 9.0

Proprietary Notice

Diamond Cut Productions, Inc. owns these software programs and their documentation. The programs and their documentation are copyrighted with all rights reserved by Diamond Cut Productions, Inc. See the License Agreement and Limited Warranty for complete information.

Published by:

Diamond Cut Productions, Inc.

P.O. Box 305,

Hibernia, NJ 07842, United States of America

These Software Products & their Documentation -

Copyright© (1993–2012) by:

Diamond Cut Productions, Inc.

P.O. Box 305,

Hibernia, NJ 07842, United States of America

All Rights Reserved

www.diamondcut.com

Notice:

Diamond Cut Productions, Inc. does not recommend the use of any of its products in emergency communications or real time intelligence gathering environments where the failure or malfunction of the product can reasonably be expected to cause compromise of the communications system, or to significantly affect its safety or effectiveness. Products are not authorized for use in such applications unless Diamond Cut Productions, Inc. receives written assurances, to its satisfaction, that: (a) the risk of injury, or damage has been minimized; (b) the user assumes all such risks; and (c) potential liability of Diamond Cut Productions Inc. is adequately protected under the circumstances.

Acknowledgements

Special Thanks Go Out to the Following Persons for their Contributions to this Diamond Cut Product:

Konstantin Themelidis, *Digital Broadcast Systems GmbH, Germany*, Gregory E. Miller, *CommEdge.com*, Ron Bowser, *Old Time Radio Collector*, Catalin Grigoras, *Ph.D., Forensic Audio expert*, Dan McDonald, John E. Ford, GB Bruncz, Tim Goodwill, Stephen E. Cook, Brian Downey, Jeff Klinedinst, *Tracer Technologies Inc.*, Curtis Crowe, *Tracer Technologies Inc.*, Denise Moyer, *Tracer Technologies Inc.*, Wesley Frank, *Albany NY*, Peter Mosher, *Ontario Centre of Forensic Sciences*, Dick Begley, *Multimedia Music, Australia*, Jason Begley, *Multimedia Music, Australia*, Carl Gerdes, Edward Noble, Douglas R. Kelly, *Ph.D., M.InfTech., Software Developer-kellyaustralia.com*, Gregg Stutchman, *Chief Forensic Analyst, Stutchman Forensic*, Jim Reames, *JBR Technology*, Marla Maier, *Diamond Cut Productions. Inc.*, Craig Borax, *Tidal Engineering, Inc.*, Les Paul, *Muscian & Inventor*, Monica Sanz Aznar, Leah Burt, Alaina Benenati, Robert Orban, Ted Nelson, *PERMANENT RECORD*, Claus Peter Gallenmiller, Joel Fritz, Art Zimmerman, *Zim Records*, Kathy Schug Elia, Steve Cain, *President - Applied Forensics Technologies International*, Mark Salomon, *Senior Systems Administrator*, Ricardo Acosta-Torres, Richard Aguila, *Project Engineer*, Dub Butler, Ryszard Parzynski and Bill Syrratt, *Soundwarp, Sydney, Australia*, Dave Rogge, *Sound Designer*, Paul Ginsberg, *Forensic Expert -Professional Audio Laboratories Inc*, Jim Kraus, *Technical Manager of Video Restoration*, John Olsson, *Forensic Linguistics Institute*, Bettina Keith, *Digital Broadcast Systems GmbH*, Chuck Ballinger, *Information Analyst*, Gina Carlson, Peter Medore, Andrew Bierwiler, Randy Torsiello, L.T. Patterson, Jos Van Dyck, Andrea Slack, *Design Engineer*, Jim Chamberlain, *Forensic Specialist – Multimedia*, Truls Birkeland, *Forensic Audio Specialist*, George Deslauriers, and Christopher Basalatan, *Graphics Designer*, Doug Carner CPP/CHS, *President - Forensic Protection* and Frank Graham, *AeroVox Forensics*.

**In Fond Memory of Les Paul for his Inspiration
and Friendship**

Table of Contents

Acknowledgements.....	4
Table of Contents.....	5
Installing and/or Upgrading Information	12
Section 1 - Welcome & the Product Basics.....	13
History of the Product.....	14
About the Product.....	15
About the Version.....	15
About the Manual	18
Getting Started.....	18
<i>Step One – Make Sure System Meets Our Requirements</i>	<i>18</i>
<i>Step Two - Installation of the Product</i>	<i>20</i>
<i>Step Three - Configure Your EIGHT/DC Forensics Program ...</i>	<i>21</i>
<i>Step Four – Connect Your Computer To Your Audio System.....</i>	<i>22</i>
<i>Step Five – Turn Screen Saver and Background Tasks Off.....</i>	<i>27</i>
<i>Step Six - Choose your Operating Mode</i>	<i>27</i>
<i>Step Seven – Testing Your System.....</i>	<i>32</i>
Step By Step Guides	35
<i>Super Easy Record Restoration Step By Step Guide.....</i>	<i>36</i>
<i>Easy Record Restoration Step By Step Guide.....</i>	<i>38</i>
<i>Advanced Record Restoration Step By Step Guide.....</i>	<i>48</i>
<i>Forensic Audio Step-by-Step Guide (DC FORENSICS Only)....</i>	<i>53</i>
<i>Live Mode Step by Step Guide</i>	<i>57</i>
Which Tool Do I Use?	61
Section 2 – The System and Its Operation.....	66
The File Menu	67
<i>Open Source.....</i>	<i>67</i>
<i>Variable Bit Rate (VBR) MP3 File Support</i>	<i>68</i>
<i>Convert Various Formats to .wav (Tutorial).....</i>	<i>69</i>
<i>Large File Conversion to .wav</i>	<i>69</i>
<i>(Expanded File Conversion).....</i>	<i>69</i>
<i>Drag and Drop File Support</i>	<i>71</i>
<i>Video/Audio Extraction System</i>	<i>71</i>
<i>Save Source.....</i>	<i>74</i>
<i>Save Source As.....</i>	<i>74</i>
<i>Data Disc Burner.....</i>	<i>75</i>
<i>DC Tune Library.....</i>	<i>80</i>
<i>Import Playlists.....</i>	<i>94</i>
<i>Close Source</i>	<i>94</i>

<i>Rip CD Tracks</i>	<i>94</i>
<i>Open Destination</i>	<i>100</i>
<i>Save Destination As.....</i>	<i>100</i>
<i>Close Destination.....</i>	<i>100</i>
<i>Clone Source.....</i>	<i>100</i>
<i>Make Destination the Source.....</i>	<i>101</i>
<i>Delete Files.....</i>	<i>101</i>
<i>Print.....</i>	<i>102</i>
<i>Print Preview</i>	<i>103</i>
<i>Print Setup</i>	<i>103</i>
<i>Page Setup</i>	<i>103</i>
<i>Exit</i>	<i>104</i>
The Edit Menu	104
<i>Undo.....</i>	<i>104</i>
<i>Copy</i>	<i>105</i>
<i>Cut.....</i>	<i>107</i>
<i>Paste</i>	<i>108</i>
<i>Append to End.....</i>	<i>108</i>
<i>Insert at Start</i>	<i>109</i>
<i>Paste Interpolate (Time Domain Technique)</i>	<i>109</i>
<i>Paste Interpolate (Bi-Modal Technique).....</i>	<i>110</i>
<i>Paste Over</i>	<i>111</i>
<i>Paste Insert</i>	<i>112</i>
<i>Paste Mix.....</i>	<i>112</i>
<i>Paste Crossfade</i>	<i>112</i>
<i>Paste As A New File.....</i>	<i>114</i>
<i>Paste Silence.....</i>	<i>114</i>
<i>Paste Bleep (tone).....</i>	<i>114</i>
<i>Select All.....</i>	<i>114</i>
<i>Pencil Editing</i>	<i>115</i>
<i>Direct Spectral Editing.....</i>	<i>115</i>
<i>Mute.....</i>	<i>120</i>
<i>Manual De-Clicking with Mute or Interpolate (Tutorial).....</i>	<i>121</i>
<i>Fade In</i>	<i>122</i>
<i>Fade Out.....</i>	<i>123</i>
<i>Single file Operations.....</i>	<i>124</i>
<i>Snap Selection to Zero Crossing.....</i>	<i>124</i>
<i>Delete All Temp Files.....</i>	<i>124</i>
<i>Gain Change.....</i>	<i>125</i>
<i>File Properties</i>	<i>126</i>

<i>Playback Controls</i>	128
<i>Play</i>	128
<i>Scrub Audio</i>	129
<i>Rewind</i>	129
<i>Pause</i>	129
<i>Fast Forward</i>	129
<i>Stop</i>	130
<i>Record</i>	130
<i>VOX Recording</i>	133
<i>Extended Recording</i>	134
<i>Play</i>	135
<i>Play Looped</i>	137
<i>Time Bracketed Play Range</i>	137
<i>Timer Recording</i>	137
<i>Make Waves Signal Generator</i>	139
<i>Change Sample Rate / Resolution</i>	142
<i>Preferences</i>	144
<i>mp3 Encoder</i>	149
<i>File Split and Recombine</i>	151
<i>Manage Presets</i>	152
The Filter Menu	154
<i>Batch File Editor</i>	154
<i>Auto Leveling</i>	157
<i>Concatenate Files</i>	157
<i>EZ Clean Filter®</i>	158
<i>Multi-Filter</i>	161
<i>Live Preview</i>	163
<i>DirectX Filters</i>	167
<i>Using DirectX Plug-ins in the Multi-Filter</i>	169
<i>Impulse Noise Filters (Clicks, Ticks, Crackle, Snaps Pops and Thuds)</i>	170
<i>EZ ImpulseNoise Filter TM</i>	170
<i>Expert Impulse Noise</i>	174
<i>Narrow Crackle Filter</i>	181
<i>Big Click Filter</i>	182
<i>Continuous Noise Filter</i>	184
<i>Harmonic Reject</i>	198
<i>Dynamic Noise Filter</i>	201
<i>Low Pass, Band Pass and High Pass IIR Filter Sub-Menu</i>	205
<i>Low Pass Filter</i>	205

<i>Band Pass Filter</i>	208
<i>High Pass Filter</i>	212
<i>Notch Filter</i>	215
<i>Median Filter</i>	218
<i>Averaging Filter</i>	222
<i>The 10 Band Graphic Equalizer</i>	224
<i>The 20 Band Graphic Equalizer</i>	226
<i>The 30 Band Graphic Equalizer (Forensics Version Only)</i>	227
<i>Paragraphic Equalizer</i>	228
<i>File Conversion</i>	230
<i>Cross Fade Filter</i>	237
<i>Virtual Phono Preamplifier (VPA TM)</i>	238
The Effects Menu	248
<i>Reverb</i>	248
<i>Echo Effect</i>	250
<i>Virtual Valve Amplifier TM</i>	253
<i>Dynamics Processor</i>	262
<i>Reverse File</i>	266
<i>Channel Blender</i>	267
<i>Punch and Crunch TM</i>	270
<i>Change Speed</i>	273
<i>Automatic Change Speed Compensator</i>	276
<i>Time Compression And Expansion (Stretch and Squish)</i>	277
<i>Filter Sweeper</i>	280
<i>Sub-harmonic Synthesizer</i>	283
<i>Overtone Synthesizer</i>	284
<i>EZ Enhancer TM</i>	286
The Forensics Menu	288
<i>EZ Forensics Filter TM (Forensics Version Only)</i>	290
<i>The Adaptive Filter (Forensics Version Only)</i>	292
<i>Brick Wall Filter</i>	296
<i>Polynomial Filter (Forensics Version Only)</i>	297
<i>Spectral Filter (Forensics Version Only)</i>	300
<i>Spectrograms</i>	309
<i>View Spectrogram</i>	310
<i>Voice ID</i>	317
<i>DeClipper</i>	322
<i>DSS Dynamic Spectral Subtraction TM (Forensics Version Only)</i>	325
<i>Cell Phone Noise Filter (Forensic Version Only)</i>	332

<i>Auto Voice Filter (Forensic Version Only)</i>	334
<i>Voice Garbler (Forensic Version Only)</i>	335
<i>Additional Forensics Features</i>	337
The Marker Menu	339
<i>Add Markers:</i>	340
<i>Clear All Markers:</i>	340
<i>Highlight Marked Area:</i>	340
<i>Drop A Marker:</i>	340
<i>Go To Next Marker:</i>	340
<i>Go To Previous Marker:</i>	340
<i>Re-Number Markers:</i>	341
<i>Label Marker:</i>	341
<i>Lock Markers:</i>	341
<i>Merge Source Markers into Destination:</i>	341
The CD Prep Menu	341
<i>Quantize for CD Audio</i>	342
<i>Chop File into Pieces</i>	342
<i>Find and Mark Silent Passages</i>	344
<i>Gain Normalize</i>	345
<i>Normalized Gain Scaling</i>	345
<i>CD Burner</i>	345
The View Menu	348
<i>The Diamond Cut Spectrum Analyzers</i>	348
<i>Spectrum Analyzer – Standard Precision (DC8)</i>	349
<i>Distortion Analyzer (THD Mode)</i>	355
<i>Spectrum Analyzer – High Precision (DC Forensics)</i>	358
<i>XY Display</i>	365
<i>Time Display</i>	369
<i>Output VU Meter</i>	370
<i>Volume Control</i>	371
<i>Fast Edit History</i>	371
<i>DC Tune Library</i>	372
<i>Zoom In</i>	373
<i>Zoom-In X2</i>	373
<i>Zoom Out</i>	375
<i>Zoom-Out X2</i>	375
<i>Zoom Out Full</i>	375
<i>Zoom to Markers</i>	375
<i>Box Zooming</i>	375
<i>Sync Files</i>	376

<i>Skins and Themes</i>	377
<i>Enabling or Disabling ToolbarsInDC8/DC FORENSICS</i>	378
<i>Channel Toolbar</i>	378
<i>File Toolbar</i>	379
<i>Filter and Effects Toolbar</i>	379
<i>Forensics Toolbar</i>	379
<i>Play Controls Toolbar</i>	380
<i>Status Bar</i>	380
<i>File Info</i>	381
<i>Rebuild Peak File</i>	381
The Window Menu	381
<i>Cascade</i>	382
<i>Tile</i>	382
<i>Arrange Icons</i>	382
<i>Close All</i>	382
<i>Open Files</i>	382
The Help Menu	382
<i>Tip of the Day</i>	383
<i>Restoring a Recording</i>	383
<i>Restoring the demo1.wav</i>	383
<i>Contents</i>	383
<i>Context Sensitive Help</i>	384
<i>Search for Help On</i>	385
<i>About DC-Art</i>	385
Section 3 - How To	386
Tutorials	386
Section 4 – Tech Support	448
Trouble Shooting	448
Reporting a Problem	458
<i>Contact Information</i>	458
Section 5 – Useful General Information	459
<i>Glossary of Terms</i>	459
Charts, Graphs and Other Info	492
<i>Additional Technical Information</i>	492
<i>Attenuation Chart</i>	493
<i>Audio Connection Standards</i>	494
<i>Audio Frequency Spectrum</i>	496
<i>Compact Discs</i>	497
<i>Decibels</i>	498
<i>Dial Tone Phone Frequency Chart</i>	499

<i>Worldwide Dial Tone Frequencies</i>	<i>499</i>
<i>Dynamic Range.....</i>	<i>500</i>
<i>Equalization Curves (Phonographic)</i>	<i>500</i>
<i>Function Finder Table.....</i>	<i>501</i>
<i>RIAA Curve Table of Values.....</i>	<i>501</i>
<i>Hard Drive Space Recording Consumption</i>	<i>512</i>
<i>Hot Key Index</i>	<i>513</i>
<i>Human Hearing Frequency Response vs. Age</i>	<i>515</i>
<i>Tape Speeds in Inches Per Second (ips).....</i>	<i>515</i>
<i>Measurement Tools Table</i>	<i>515</i>
<i>Rotary Head Tape Recorder Speeds:</i>	<i>518</i>
<i>Musical Scale.....</i>	<i>519</i>
<i>Resistor Color Code</i>	<i>520</i>
<i>Sound Level.....</i>	<i>520</i>
<i>Stroboscope.....</i>	<i>521</i>
<i>Sync Mode/Non Sync Mode Explanation Chart</i>	<i>522</i>
<i>Turnover Frequency Chart.....</i>	<i>523</i>
<i>Wire Table</i>	<i>524</i>
<i>Equalization Chart for LP Records</i>	<i>525</i>
<i>Language Translation (Deutsch/Español)</i>	<i>527</i>
<i>Preset Listings.....</i>	<i>533</i>
<i>A Brief History of Diamond Cut Productions.....</i>	<i>547</i>
<i>Diamond Cut Audio Restoration Tools Development Timeline</i>	<i>549</i>
<i>Diamond Cut Productions Edison Lateral Series CD and Cassette Releases</i>	<i>553</i>
<i>DCAT-3 Audio Test CD Set.....</i>	<i>555</i>
<i>Tracer Technologies ...For the Supplies You Need.....</i>	<i>557</i>
<i>Diamond Cut Software Product Model Number Nomenclature (English Versions).....</i>	<i>558</i>
<i>License Agreement</i>	<i>559</i>
<i>Index.....</i>	<i>561</i>

Installing and/or Upgrading Information

Here is some important information to be aware of when installing your new Diamond Cut version 8 software. If you are upgrading from an earlier version of a Diamond Cut Audio Restoration product such as DC7, there are some things to be particularly aware of.

1. If you've purchased a hard copy version of the product, you will find your serial number located behind the install disc which is contained in a pouch located at the back of the printed user's manual.
2. If you have purchased a new installation (not at upgrade), you will need to enter a user name, a serial number and your email address. After that, the software will fetch your registration code via your internet connection or via email. If you do not have an internet connection, you can call us or write to us to obtain a registration code. We will need the exact user name that you used when you installed the software and your product serial number. If either of these are not provided exactly as entered into the software, the system will not register and, will not work after the time trial period has expired.
3. If you purchased an upgrade, the Diamond Cut system will look for a previous registration in terms of your User Name, Serial Number and/or email address. The system looks for existing Millennium, DC6, DC7 or DC8 registrations. If it finds one, it will extract the required information and use it for the electronic registration of your upgraded product which should occur within 24 hours.
4. After product installation has been completed, please check for software updates at www.diamondcut.com because we are in the continual process of creating product improvements and bug fixes.
5. It is a good idea to check for free product updates ever few months to keep your software current in terms of bug fixes and feature enhancements.
6. You do not need to un-install DC7 before installing DC8 or DC Forensics Audio Laboratory, v.8. As a matter of fact, it may be of advantage to leave your earlier version of DCArt (Diamond Cut Audio Restoration Tools) installed if they contain important presets that you

have created over time or if you want to maintain your existing DCTune Database and playlists.

7. You do need to un-install previous versions within a product family before updating. For example, if you are updating your software from DC8, version 8.03 to version 8.1, we recommend uninstalling 8.03 first.

8. If you have personal favorite presets that you created using an earlier version of DCart, you can use the Preset Manager in DC8 to import them into the new preset directory created by DC8.

9. The first time that you open your new DC8 DCTune Library, the system will ask you if you want to import your old DCTune Database from the DC7 directory into your DC8 directory. If you choose to do this, it will also transfer your old playlists into the DC8 Directory.

10. It is a good idea to record your User Name, Serial Number and Registration code for safe keeping in case your hard drive fails. A good place to do this is in the “Notes” section of the printed users guide. You will need that information to restore your system in the event of a system failure. To find that information, go to your Diamond Cut Help\About--- menu.

Section 1 - Welcome & the Product Basics

Thank you for purchasing another fine product from Diamond Cut Productions, Inc. We strive to provide you with audio DSP software that push the envelope in terms of features, but leave enough money in your wallet to afford the envelope. We will do our best to not only provide you with the most effective DSP audio software available anywhere, but also complement it with excellent support materials. This manual is your window to the product. Please use it. A great supplemental source of information can be found at the Diamond Cut Productions forum located at:

<http://www.diamondcut.com/vforum/index.php>

It is a free service and has a large body of information built up over a long period of time by users of the various Diamond Cut products.

Though we will support this product to the best of our ability via telephone, email, and our web site, chances are, everything you need to navigate EIGHT and FORENSICS 8 is right here. We've done our best to make this manual and the other supporting documents as user friendly and intuitive as possible...give it a try... "reading is fundamental".

History of the Product

Diamond Cut Audio Restoration Software was originally written by two engineers in their spare time to facilitate the very specific needs that arose in their restoration of the Edison Lateral Collection of Test Pressing Recordings, which is located at the Edison National Historic Site in West Orange, New Jersey. Rick Carlson and Craig Maier developed this program over a 19 year period of time. When they made it available to the general public many years ago, it was with the idea in mind that if it solved some audio restoration problems for them in dealing with the Edison archive, it might also be of use to others confronted with similar audio signal processing problems. Since then, it has grown into the most used audio restoration and enhancement product on the planet, and beyond.

How to Get Started

This manual is nice and thick but it's not a novel and we don't expect you to read it cover to cover. However, we'd like you to get started by carefully reading Sections 2 and 3. Section 2 covers how to install the software, hook up your external sound system, and test your installation. Section 3 gives you some great hands-on tutorials that take you step by step through some real restorations. If you go through both of these sections, you'll be fully oriented and ready to use the program!

Important Note: This manual supports 2 different products. As many of the features for both DC EIGHT and Forensics Audio Laboratory v.8.1 are similar, we have streamlined this manual to support both products. As you browse the Table of Contents, you'll notice several entries marked "Forensics Only". These will only be supported in DC Forensics Audio Laboratory v. 8.1.

About the Product

DC EIGHT and DCForensics Audio Laboratory v.8.1 contains a comprehensive set of tools designed for audio restoration, enhancement audio archiving and audio surveillance. This extensive toolkit will allow the user to remove extraneous noise and also enhance the sound from any audio source without degrading the content contained on the original. Recognizing that there is a tradeoff between the degree of noise removed from a source and the fidelity, transient and frequency response maintained, we have sought to provide the highest level of user control while maintaining ease of use over the variables that affect the audio restoration and enhancement process.

Professional Audio Engineers, Forensic Scientists, Audio Archivists, Ham Radio Operators, and Audiophile Hobbyists use our products for re-mastering, editing, noise removal and audio signal analysis. It is also used by many local, state, and federal government agencies including military intelligence operations for forensics law enforcement applications, particularly in surveillance situations. Private sector forensics audio engineers use it for post processing forensic recordings of phone calls and surveillance tapes. Radio and television networks have found the LIVE (real-time signal processing) feature of particular value in their real-time applications. Additionally, a number of Engineering Colleges and Universities use this software as part of the laboratory portion of some coursework pertaining to applied DSP techniques.

Important Note: Most of the algorithms used in DC EIGHT and DC DCForensics Audio Laboratory v.8 use double precision floating point math as opposed to fixed precision integer math in order to minimize the possibility of introducing digital noise into your .wav files. The tradeoff associated with using this method is the time required to process a file being slightly longer than the fixed precision integer method.

About the Version

DC 8 (v.8.1) and DC Forensics Audio Laboratory (v.8.1) have been in the works for over a year. We've added a ton of novel new features. If

you've been using our products and are only new to DC EIGHT or DCForensics Audio Laboratory v.8.1, use the list below as your table of contents. It includes every new feature to the product so that you can quickly access all of the newest info without rehashing the stuff you've already read. Here is a list of the features making their debut in Version 8.1:

DC Eight & DC Forensics V. 8.1 New Features:

- Big Click Filter Added
- Big Click Checkbox added to the EZ Impulse Filter
- Direct Spectral Editing mode (DSE) added to the Spectrogram
- Manual Click replacement provided in the Spectrograph View
- Extended the length of the time interval capability of the Manual Interpolation Function
- Added a bi-modal algorithm to the Manual Interpolator which uses both time and frequency domain techniques for greater signal replacement accuracy
- Independent Hot key access to the Time Domain Only manual Interpolator
- Added a Sub-Harmonic Synthesizer Effect for recovering lost lower Octaves of the audio spectrum
- Added an Overtone Synthesizer Effect for recovering lost upper Octaves of the audio spectrum
- Provided Multifilter support to the DCTune Library
- Added Continuous Play Mode to the DC Tune Library
- CD Player Added to the DCTune Library
- Spacebar now starts and stops the play of any file in the DCTune Library
- Time Display added to the DCTune Library
- DCTune Library has an expanded list of default Genres
- Broadcast .wav (BWF) file support added
- .wav file header editor added
- Vorbis (Ogg Vorbis) Lossy Compression File Support added
- FLAC (.flac) Non-Lossy File Compression File Support Added
- Added a Shuffle Play Feature to the DC Tune Library
- Presets added to the Make Waves Signal Generator

- Entire Musical Scale Ranging from C0 to D9# added as Presets
- Data Disc Burning Capability for CDs, DVDs & CDR-Ws
- Disc at once mode added to the CD burner
- CD text support added to the CD burner
- Data Disc Burning Capability Added (CD & DVD)
- Hotkey access added for the Spectrogram Function
- Added 49 LP Phono Equalization Curves to the VPA
- Added numerous presets to other filters and effects
- Marker Transfer capability provided between the Source and Destination Windows
- Scrubbing Tool added to rock back and forth over a file sector
- Time Domain Tracking / Waveform Scrolling while playing is provided
- DC Playlist play bar scrolling and random access to play position provided
- User Selectable Display Themes provides choices of skins and color schemes based on personal preference
- Improved the flexibility of the Spectrogram Function
- Increased the Frequency Resolution of the Standard Spectrogram
- Added Gamma Control and Inverse Palette to Spectrogram
- Normalized Gain Scaling added to the Batch Processor
- Expanded Play List extension(s) support
- Secondary Sort Capability added to the DCTune Library
- Mouse Wheel controlled Zooming In & Zoom Out
- Improved De-Clipper (Forensic Version Only)
- Voice ID Feature Added (Forensic Version Only)
- Cell Phone Noise Filter Added (Forensic Version Only)
- Voice Garbler / Disguiser (Forensic Version Only)
- Auto Voice Filter Added (Forensic Version Only)
- File Systems directories compliant with Win 7 requirements (Forensic Version Only). See the Appendix section for the location of Presets and demo wavefiles.
- Added Shortcuts on the Start Menu for all of the demo wave files.
- Added an IIR option to the Harmonic Reject Filter
- Fast-Edit Temp Files kept in same directory as the Source File

About the Manual

With roughly 575 pages of information passed on to us by our scientists and engineers, we tried to format a manual that was basic enough for a first time, non-experienced user, but filled with the kind of detail that any “propeller head” would love. We’ve also condensed a good deal of the material...all that means to you is that if you have an interest in a specific item such as the Punch and Crunch Filter, you simply look it up in the Table of Contents or Index. You’ll not only find all the information describing the tool, but also any tutorial information on that specific tool all in the same location. Our Tutorial section now contains only multiple tool procedures. As mentioned earlier, this manual supports both DC Forensics 8.1 and EIGHT, so be aware that features native only to the Forensics version will be marked accordingly. Information contained in the product’s help file will sometimes supersede information contained in the printed Users Guide, because the software (and thus its help file) is updated more frequently than the printed material.

Getting Started

Note: Everything you need to know to install the software, hook up your hardware and begin to use the program will be found in the next two chapters. This is *must* reading. You’ll find it easy and fun!

So, enough of our yacking. Let’s install this software and have some fun.

In this section, we’ll tell you everything you need to know in order to install, configure, check out and start using the software. Read through this section carefully and perform the steps indicated and you’ll have a perfectly working real time audio restoration workstation!

Step One – Make Sure System Meets Our Requirements

Most machines today are more than capable of handling audio processing. Hard drives have also grown to a point, where you really have no limits on what you can do...except perhaps time. As we’ve not yet developed time manipulation outside of the computer, let’s just

deal with what you'll need in the way of a PC. Here are the minimum System Requirements:

- 750 MHz Pentium or better (Previewing multiple filters strung together in the Multi-Filter may require faster processors.)
- 16 bit (or better) Stereo Sound Card with line level inputs. Real time feed through requires a full duplex sound card that can record and play at the same time. *
- 1024 Mbytes of memory (RAM) for XP, 2GB for Vista or Win7**
- Windows XP with SP3 upgrade, *** Vista or Windows 7 (32 or 64bit)
- Audio Source Material
- An Audio Delivery System (to match whatever source recordings you have)
- A Hard Drive with enough space to accommodate your .wav files. A formula is provided to calculate the space requirement but you should have about 2.5Gbytes free to make a full audio CD. (The program itself only requires about 30 to 45 Mbytes of space, depending on the version.)
- Mouse, Keyboard, and Color Monitor

* You now possess space age technology for audio restoration and enhancement. If you haven't yet replaced the sound card that came with your computer (which is probably the lowest quality audio component you own), contact Tracer Technologies or visit them at www.tracertek.com

** If your computer has more than the minimum required to run your OS adequately, further increases in the quantity of RAM will not appreciably speed up the DC 8 / DC Forensics 8 algorithms since they are almost totally limited by the processor. Upgrading to a faster processor WILL allow the filters to run much faster. Most DC EIGHT/DC Forensics 8 algorithms use a single thread and will not be appreciably speed up by multi-core or multiprocessor CPUs.

***This product uses DirectX and requires DirectX 9.0 or higher. Some older computers may require additional component installations.

Step Two - Installation of the Product

Most users find this extremely easy.

1. Put the install CD into your drive. If Auto-run is enabled, installation will begin immediately. If not:
2. Click the “Start” Button, choose “Run” from the pop-up menu
3. Click the “Browse” Button and point to your CD ROM Drive
4. Find the dcautorun.exe icon and double click on it
5. The install program will start. Follow the instructions on the screen.

The software will be installed in a folder called *Diamond Cut Productions*. Several demo .wav files are also supplied as part of the demonstration tutorials in both the Help file and PDF version of the manual.

During the installation process, you will be asked to approve the license agreement, and select an install location. After a few seconds of installation time, you’re ready to launch the program by clicking on the EIGHT or Forensics 8 Icons, depending on which program you have purchased.

Note: DC EIGHT and DC Forensics 8 require a serial number and registration with Diamond Cut Productions, Inc.

When DC EIGHT or Forensics 8 is first run you must choose your User Name and enter it into the software. You also will need to enter the Serial Number that was supplied to you when you purchased the product. Enter the serial number exactly as shown using only the numbers 0-9 and letters A-F.

The combination of your name and serial number is your unique key to EIGHT or DC Forensics, no other personal information is used.

After you have entered your name and Serial Number, you must then register the program with Diamond Cut Productions. The registration process can be done automatically over the internet or via phone or email.

If you choose automatic registration, DC EIGHT/DC Forensics will attempt to contact the servers at Diamond Cut Productions and automatically register your product. The registration process installs a registration code into your computer and activates the product. After

registration you will receive an email with a duplication registration code for your records, should you ever need to re-install the product.

If you register by phone or email, you will receive a registration code from Diamond Cut Productions that you must enter into the program to complete the registration. We strongly encourage either automatic or email registration. .

Note 1: Diamond Cut Productions is the ONLY authorized source of registration codes, regardless of where you purchased your product.

Note 2: It is a good idea to record your User Name and Serial Number someplace easily found in case your hard drive fails causing you to have to re-install the software. A page called “Notes” is allocated at the back of the printed version of the User Guide for that purpose. Your registration code can be found in the Help-About dialog box.

Step Three - Configure Your EIGHT/DC Forensics Program

EIGHT/Forensics 8 does not require much configuration, but it does require that your sound card be installed and working properly. If you have more than one sound card in your system, make sure the one you wish to use has been selected in the Sound Card Selection screen (in the Edit/Preferences/Soundcard Menu).

Next, check the Temp File Path under the Edit/Preferences/General Menu. EIGHT/DC Forensics 8 automatically assigns temporary file names for files that are being processed. You should set this temporary drive path to the disk drive that you wish to use for audio editing. This is usually the drive with the most free disk space. Keep in mind that high quality (44.1 kHz) stereo recording consumes 10.5 Mbyte of disk space per minute.

Generally, this is where we get calls from customers who experience one form of problem or another. This is usually because most people don't record onto their computer's hard drive very often and they may not have their system set up for this type of application. We're happy to help, but before you call us... please read the next few paragraphs:

1. Make sure you have the output of your audio source plugged directly to the “Line” input of your sound card. (Do Not Use the Microphone (Mic) Input of your sound card).
2. If you have a turntable, make sure that you are using either a stereo preamp or the “Tape Out” on your Stereo system in between the sound card and the turntable. Your turntable cannot generate enough signal level on its own to make a good line level recording. You’ll need the additional amplification to boost your turntables signal to be compatible with the “Line” input of your sound card.
3. If you try to play or record and you get the message “Cannot play the specified format...etc.” This is a message generated by Windows...not our product. It is telling you that EIGHT/DC Forensics is trying to play a .wav file through a device that is not able to do this. This could be a result of Windows having a Modem listed as the primary device for playing .wav files...or some other non .wav device. Check in your Start Menu/Control panel settings to make sure that your sound card is listed as the primary playback device. Also, make sure EIGHT/DC Forensics is set correctly for your sound card under the Edit/Preferences/Soundcard menu.
4. If you can hear audio playing through the speakers but the record VU meters are not jumping, then check your sound card’s mixer. Many times, the recording mixer is turned down or muted. Just click on the mixer (in many cases it’s a little yellow speaker icon located in your Windows Tray beside your clock), and go to Options/Properties and look for “Recording”. Click on “Recording”, then click “OK” to select your input mixer so you can turn up or Select your Line inputs. Important note – also make sure your “Line” input is selected in this mixer – not your “Mic” input.

We’ll cover more in the Troubleshooting section of the manual, but these are the highlights.

Step Four – Connect Your Computer To Your Audio System

Your sound card has both an input and an output. The output is used when you play an audio file and the input is used when you record audio into the computer. Refer to your sound card manual or the little

icons on your soundcard to identify which jack is for output (playing back audio) and which is for input (recording audio). Often, a small speaker or headphone icon represents the output jack.

We first want to connect speakers to the output of your sound card so we can hear audio. You probably already have speakers connected. If you do, then fine. Just leave them alone. If not, you'll need to connect the output of your sound card to either an amplifier or amplified speakers with line level inputs.

Note: Your sound card records and plays audio at a certain established Voltage level. This standard level is referred to as Line Level. Just about all devices play or record at this exact audio level so they're all compatible with each other. CD players, Tape Recorders, Amplifiers, Receivers, VCRs, DAT machines, Mini discs and most other devices all play or record this standard Line Level signal. This means you can plug your tape player output or other line level device directly into the input of your sound card since they both use the same type level of signal.

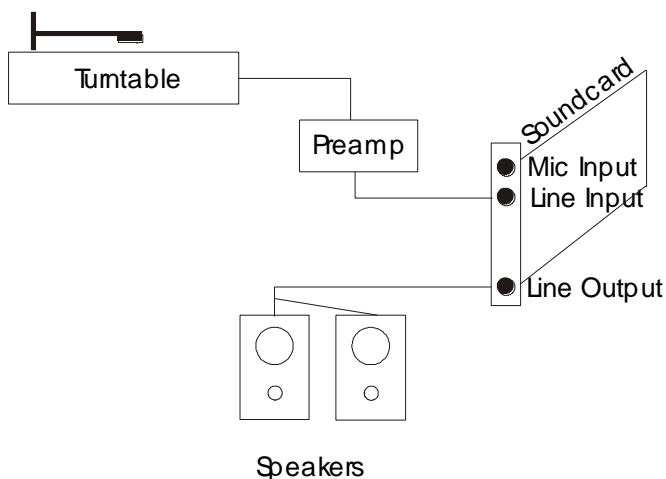


Figure 1 - Wiring Things Up!

Next, we need to plug some audio into the input of your sound card so we can record this audio into your computer. If you are using a line level device, just plug it into the sound card "Line" level input using

whatever adapter cable mates your player device (i.e., tape recorder) to the input jack on your sound card.

Critical Note on Turntables - There are two common playback devices that are not line level and which can't be plugged directly into your sound card line level input. The first is a turntable with a magnetic phono cartridge. A magnetic phono cartridge produces a signal that is much lower in strength than line level. Your sound card cannot use this signal as it is – it must be amplified up to the standard line level before it gets to your sound card.

If your turntable has a magnetic phono cartridge, you must use what is called a magnetic phono preamplifier. If you have a stereo system with an input jack on the back-labeled “Phono”, you already have a preamp. It's in your stereo system. Just leave your turntable connected to the “Phono-Input” of your stereo and connect a Line Level output of your stereo to the sound card input. Many times, this line level output is marked as “Tape Out”. Now, any audio (including your turntable) you can hear from your stereo system will be available to your computer for recording. Another option to consider is the use of a “Flat Phono Preamplifier”. Flat preamplifier front ends, when used in conjunction with the Diamond Cut Virtual Phono Preamplifier (VPA™) provides a high level of EQ curve flexibility as well improved sonic performance of your transfers. For more information on this technique, you can download an Application Note from our website located at:

<http://www.diamondcut.com>

Navigate to the App Note section of the site and look for AN 8 titled “The New Way of Recording LPs”.

If you don't have a stereo system to connect your turntable to, or it's in another room, you'll need a stand-alone preamplifier. These are not expensive and you can get one at www.tracertek.com or www.diamondcut.com

Again, this device takes the tiny signal from your turntable and amplifies it so that it's now a standard Line Level signal that can be plugged directly into your sound card.

Note 1: Low cost turntables utilizing Ceramic Phono Cartridges do not require a special pre-amplifier in that their signal level is already at approximately line level.

Note 2: Be sure to connect the turntable-grounding wire (strap) to the chassis ground of your Phono Preamplifier or to the computer chassis itself to minimize hum.

Another device that does not develop a standard Line Level signal is a microphone. If you want to record from a “mic”, just plug it into the jack on your sound card that is labeled “Microphone”. If you need to use this mic jack, again, refer to your sound card manual for specifics.

Note: Some laptops have only two analog audio jacks. One is always an earphone / audio output jack while the other is often labeled “mic”. Many laptop computers allow you to convert the functionality of this “mic” input over to that of a “line” input via the sound system chipsets driver routine control panel. Please refer to the manufacturer of the laptop for details on this mic/line level switchover functionality.

In most cases, you’ll simply be taking the output of your source to the input of your sound card. This is pretty easy to visualize since the computer is really acting like a familiar tape recorder with an input and an output. Many audiophiles have very sophisticated and complicated audio systems that allow many different kinds of hookups. Therefore, there are many alternative methods for connecting your computer to a sound system in order to be able to use EIGHT/DC Forensics. Here are several methods: (NOTE: Average users will not have to use these or other complicated hookups – just supply an audio signal to the input of the sound card and listen to it on the output.)

Method #1: Using a home stereo tape-monitoring loop

1. Connect a stereophonic magnetic phono pickup system to an audio pre-amplifier with magnetic phono equalization inputs.
2. Connect your line level sound card input to one of the pre-amplifier's tape recording outputs.
3. Connect your line level sound card output to one of the pre-amplifiers tape monitoring inputs.

Method #2: Using a DAT with digital (S/PDIF) inputs and outputs

1. Connect a stereophonic magnetic phono pickup system to an audio pre-amplifier with magnetic phono equalization inputs.
2. Connect the DAT machine analog output to a tape monitoring input on the pre-amplifier.
3. Connect the DAT machine analog input to a tape output on the pre-amplifier.
4. Connect the Digital Output of the DAT machine to the Digital Input of a "Digital-Only" sound card in your computer.
5. Connect the Digital Input of the DAT machine to the Digital Output of the "Digital-Only" sound card.

Method #3: Using a mixing board and Analog Sound card

1. Connect a stereophonic magnetic phono pickup system to a magnetic audio pre-amplifier (these are available without all of the bells and whistles associated with a full-blown home audio pre-amplifier).
2. Connect the Outputs of the magnetic pre-amplifier to two of the line level inputs on your mixing board (one input for each channel).
3. Connect the line level outputs of the analog sound card to another pair of line level inputs on your mixing board.
4. Connect your tape recorder (DAT or Reel to Reel or whatever) line level outputs to another pair of line level inputs on your mixing board.
5. Connect any other input devices you may require into the remaining inputs of your mixing board.
6. Connect the Main mixer output to your power amplification system.
7. Connect the Monitor Outputs from your mixer to the line level input of your sound card.
8. Connect the tape recorder line level input to the Stereophonic Headphones output jack on your mixing board.

Warning: Method #3 is the most versatile method for setting up a small sound restoration lab. However, because it is so versatile, feedback loops are easily created which can produce very annoying and potentially dangerous signal levels (to your ears, power amplifier and

loudspeakers). So you must be careful not to allow the output of a device to feedback into the same device when operating the mixing board. Always check twice before raising a slider control on your mixing board utilizing this method.

Step Five – Turn Screen Saver and Background Tasks Off

In some cases when recording, if your screen saver or other background computer task automatically comes on, it may interrupt your recording or add glitches that you didn't want. It is always better to turn these off before recording. Though this is less of a problem with faster computers, most users will feel more secure if they turn off automatic backups, screen savers, etc. This allows the computer to fully concentrate on recording clean audio.

Step Six - Choose your Operating Mode

The software can be operated in either of two basic operational modes. Choose your operating mode based on your personal preferences. Beginners will likely find that the Fast-Edit mode is easier to get oriented with since it works more like traditional audio editing programs. More advanced users will likely switch to the Classic mode since this offers a fully optimized restoration environment.

Fast-Edit Mode

The **Fast-Edit** (single file editing mode) mode operates much like a word processor where all editing is done on one file. The original file is not modified until a Save is performed. Fast-Edit mode maintains a separate history file representing the editing sequence and offers almost unlimited Undo capability. The advantage of the Fast Edit's Single file editing mode is its "greased lightning" speed, leaving you with more time for your domestic chores. Where editing examples are used in this manual in order to highlight the use of an editing or filtering sequence, the **Classic Edit** (Source and Destination) mode is utilized.

In this mode, you preview the processing results, and if not satisfactory, you can use the "Undo" feature found in the Edit menu. Also, you can highlight a particular step in the **Fast-Edit** history in order to quickly

jump back to a previous editing state. The editing processes will be temporarily undone back to the selected point in the Edit History Monitor. If you want to permanently go back to a previous editing state you can simply double click on the last edit you want to delete in the **Fast-Edit** Window. All editing done after that point will be removed and you can continue your editing session. The Delete function can also be found by clicking with the right mouse button. Fast Edit temp files are maintained in the same directory as the source file and include elements of the original file name for ease of identification.

Classic (Edit) Mode

EIGHT/DC Forensics **Classic (Edit)** mode usually operates on files in a non-destructive manner. The Source and Destination mode (the Classic technique) involves the use of a “Source and Destination” set of files. When a file is processed with a DC EIGHT/DC Forensics filter or effect, the software reads the Source file, modifies it with the selected filter or effect, which then writes it to the Destination file. The main workspace of EIGHT/DC Forensics 8 always has a Source and a Destination file in the **Classic (Edit)** mode. This mode of operation has a few important benefits:

1. The original source file is not modified, leaving it available for instant comparisons with the processed version.
2. The original material can always be recovered if the results of processing are unsatisfactory.
3. Selected sections of the file can be reprocessed using different filter parameters or different filters entirely (see sync mode).
4. Every filter that is run results in a new file which can have yet another filter run on it. All these intermediate files are always available so the users can instantly go back one or more steps in the restoration process.

The Source and Destination Workspace

When you open a .wav file in EIGHT/DC Forensic’s Classic Mode, two workspaces will appear. The top one, called the Source Workspace will display an envelope consisting of the program peaks of the .wav file just opened. If you are using Peak files, the entire waveform should be visible in the window. If you have peak files turned off, then

the amount of the .wav file that will be displayed is determined by your "display limit" settings. Both settings can be found in the Edit/Preferences/General section of the Edit Menu. The display will consist of a black signal on a yellow background.

The Destination Workspace just below the Source workspace will contain no waveform information initially, and will contain a gray background color. Both of these two workspaces display amplitude on the Y-Axis (vertical) and time on the X-Axis (horizontal). When you initially open a file, the entire file is displayed, and is periodically represented by a sample of the peak of the waveform envelope. When you Zoom-in on a portion of the waveform, at some value of magnification, you will begin to see continuous waveforms, rather than impulse representations of your .wav file signal. For more information regarding Zooming-In on a .wav file or Zooming-Out on a .wav file, please refer to the sections entitled "Zooming-In & Zooming-Out on portions of a .wav file." Please note that the active workspace is always shown in yellow.

If you are working with stereo .wav files, the workspace will display a pair of waveforms. The top waveform in either of the workspaces represents the left channel while the bottom waveform represents the right channel. If the signal is monophonic, only one waveform will be seen in the workspace(s).

At the top of the DC8 screen is a Title bar, which contains the name of the opened Source .wav file. At the bottom of the DC8 screen and on the right side, you will see five little boxes. The first shows the Mode in which the file was recorded or processed ("Stereo" or "Mono") followed by the Sample Rate that was used to create the file and then the Bit Depth. The fourth box give the total running time of the Source .wav file and the final box shows the space remaining on the hard drive being used.

After a .wav file has been processed by one of the functions under the Filter command, the output of that file will be sampled just like the Source file and displayed in the Destination workspace just below the Source workspace. It will become highlighted in yellow just following the completion of a processing session.

At the bottom of each of the two workspaces, you will see several time displays. Each display is indicated in Minutes: Seconds: Milliseconds. The time display on the left side of the workspaces indicates the starting time of the portion of the .wav file being displayed in the particular workspace. The time display on the right side of the two workspaces indicates the ending time of the displayed portion of the highlighted .wav file. When a file is initially opened, the display on the left will indicate 00: 00: 00. The right display will indicate the total time duration of the opened .wav file. If you use the Zoom function, the left display will now display the start time of the highlighted Zoom-In portion of the .wav file. The right time display will indicate the end-time of the highlighted Zoomed-In portion of the .wav file. The total time duration of the Zoomed-In highlighted portion of the .wav file will be displayed on the status bar located below the workspaces.

At the right hand side of each of the two workspaces, you will see two vertically oriented slider controls next to one another. These are useful for viewing details in a selected portion of a waveform which has been Zoomed-In on. For example, there may be a small transient that you want to see in more detail that is riding on top of a much larger waveform. The control on the farthest right is the “Display Gain” control. Using your mouse, you can move this control up and down in order to change the display gain. Moving it down will increase the gain of the display, making the waveform larger on the display (you can also adjust the Display Gain using your mouse scroll wheel). However, it may get so large as to move the portion of the waveform in which you are interested off of the top or the bottom of the display screen. The control just to the left of the gain control is the “Offset” control; this is used to move the entire portion of the waveform in which you are interested back into view. You should experiment with these controls a few times to get a feel for how they behave, and then you will begin to understand their usefulness.

At the bottom of each workspace, you will see the “Time Axis Scroll-bar”. This control is also operated by the left mouse button, and is used to move the “Play Pointer” to various locations within the display workspace. Sometimes, there can be a few seconds of delay when using this slider, so be patient as it performs the calculations to keep up with your commands. When you are Zoomed-In on a portion of a file, the

slider control can be used to move the display start-point within the highlighted field using either the slider with your mouse, or by using the arrow controls which are located at each end of the slider. The Time Axis Scroll-bar position is always relative to the entire file length, no matter how zoomed-in on a particular waveform you may be. Clicking on the right arrow button will move the waveform to the left of the workspace 1/10th of the overall display length and clicking on the left-hand arrow button will do the same thing, only moving the file in the opposite direction. If you click on the Scroll-bar (not the slider control itself), the waveform will move one full frame to the left each time you click.

Note 1: The Time Axis Scroll-bar is inactive when you are fully zoomed-out.

Note 2: If you are not using Peak Files, then the software only reads the first few Megabytes of a .wav file for the initial display. No .wav file processing operations are adversely affected by this action. Portions of your .wav file not shown on the display can still be played, filtered and operated on just as if they were displayed. To set the size of the waveform that will be displayed, use the Preferences dialog box (found under the edit menu) and increase the "Display Length Limit" to the size of the file you wish to be displayed. Keep in mind that the larger the display size, the longer it will take to initially open a .wav file. We recommend the use of Peak Files for faster display.

Note 3: The Destination Workspace can be converted into an Audio Spectrograph time synced to the Source Workspace. Please refer to the appropriate section of the manual for details on the operation of the Audio Spectrograph/Spectrogram.

Note 4: How to Choose Your Mode - Switching between **Classic** and **Fast Editing** mode is fairly simple. If you start with one and decide you want to try the other one, you can simply go to the Edit Menu and click on Preferences and the General Tab. There you will see a check box that enables Fast Edit Mode. You then need to exit the program and re-run it for the changes to take place or you need to close any open file and then re-open it for the new mode to be invoked.

Step Seven – Testing Your System

By now you should have installed the software and connected your speakers to the sound card output and some audio playback device (such as a tape player) to the sound card input. It's time to make sure your efforts have borne fruit.

The first thing we're going to do is run the EIGHT/DC Forensics program. Do this by double clicking the new DC8 icon on your desktop. Put away the tips screen and click on File/Open Source. You'll now see a familiar Windows file selector box. Navigate to C:/program files/Diamond Cut Productions/DC8/Wavefiles (or wherever else you chose to install DC8). If you are using Forensics 8, the demo files are located as shortcuts in the Start menu. Open the file called Demo1. This is a standard .wav file and was included with the program. We'll actually clean up this file in the next section, but for now we just want to play it.

Once the file is selected, it will open in the program and you'll see the graphic drawing of the audio waveform on your display. It will look something like this:

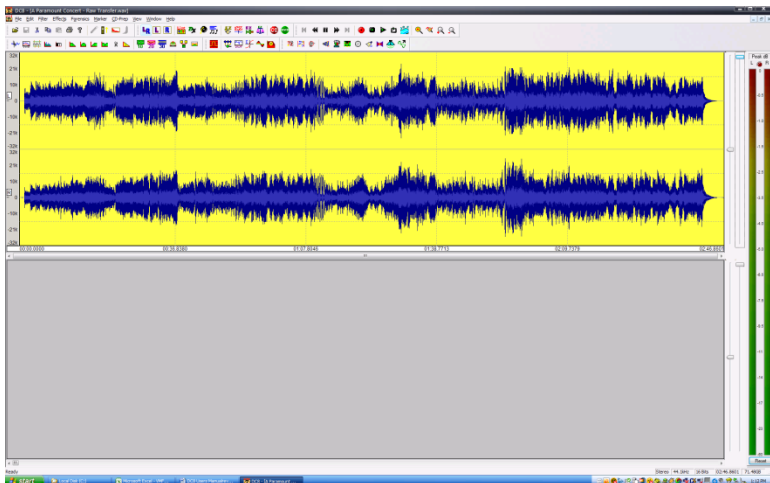


Figure 2 - A File is Opened

Notice all the nice icons at the top of the display. If you move your mouse over any one of them and leave it there for a second, a little box will open telling you what that icon does. To play our file, we need to find the little icon that looks like a right pointing triangle. This is the standard symbol for “play” that you’ll find on any tape deck. Click on this icon and you should hear audio and see the cursor move.

Note: If you aren’t getting audio at this point, check out our Troubleshooting steps found later in the manual.

Let the audio play to the end of the file and it will stop automatically and reset itself so you can play it again. Let’s play it again using a keyboard shortcut. Shortcuts are simply keys on your keyboard that perform a function without having to use the mouse. Power users love shortcuts and the Play shortcut is the easiest of all. Just hit the spacebar now and the audio starts playing. Hit the spacebar again to stop the audio from playing. Easy, isn’t it? Many beginners like to spend hours doing this, but let’s move on to test the recording capability of your system.

For our recording test, we’re going to assume you have a tape recorder connected to the input of your sound card. A CD player or even a turntable with a preamp uses exactly the same process, so just make sure you’re hooked up correctly.

To record, start your tape recorder playing a pre-recorded tape (or start playing an LP, etc). You should hear the audio coming from your speakers. Now click the red Record button on the toolbar in DC EIGHT / DC Forensics. A nice record screen comes up which looks like this:

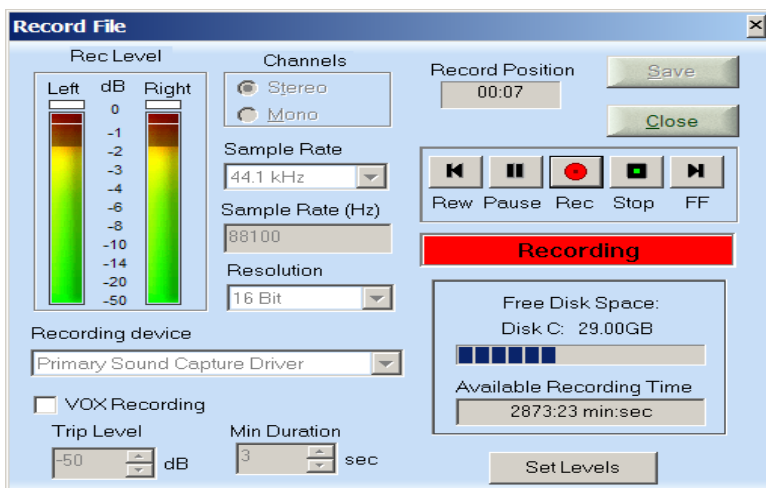


Figure 3 - The Record Window

This screen will allow you to set the recording parameters and make sure the recording is happening correctly. Let's start by making sure we are set to record a stereo file at 44.1 kHz, 16 bit. Just use the drop down boxes to select these parameters. Note the Recording Device box. Your sound card should be listed here. If not, drop the box down and choose the sound card that has the audio being fed to it. If you click on the "Set Levels" button, the following drop-down will appear which will allow you access to the Windows Mixer controls.

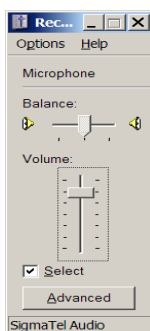


Figure 4 - Windows Sound Card Mixer Control

Now click the Pause button in the Record Window. This puts the program into what is called Record/Pause mode. The VU meters on the left should now start dancing as the program is actually seeing the audio from your tape recorder.

If those meters are dancing, adjust the input level until they are reasonable (not too high or clipping and not too low) and you are now done with testing your installation. Click Stop, and then click Close. Answer “yes” when the program asks if you want to discard this recording. You now have a perfectly working audio restoration workstation and you can go on to the next section where we’ll teach you to use this powerful tool.

If you found that your meters didn’t jump when you hit the Pause button, the problem is very likely caused by your sound card being set to the Mic input and not the Line Level input. Full troubleshooting help is included later in this manual, but this common problem is easily and permanently solved by double clicking the little yellow speaker icon in your Window tray – just to the left of the clock on the bottom of your screen. Now choose Options and then Properties. Click the record button and then click OK. Now click the check box under Line Input and your meters will start to jump in EIGHT/DC Forensics indicating that all is now well. You just turned on the Line Level input – which is where our audio is, after all.

Now that your system has been installed and checked, it’s time to get to the software and perform an actual restoration project.

Step By Step Guides

In this section, we’re going to take you by the hand and lead you through some complete examples of typical restoration projects. You’ll remove noise and “sweeten” the audio. When you are done, you’ll be licensed to drive the product yourself without training wheels.

This entire training process will take you about 10 minutes and will give you all the basics you need to know to use the program. Please follow along step by step thru the first three examples – Super Easy, Easy and Advanced Record Restoration. If you are a Forensics audio user, please go thru all three Step By Step Guides.

In these Guides, we'll tell you exactly what to do and even explain why it is you are doing these things. It's important that you follow along exactly as one step builds upon the next.

Note: The actions you are expected to take are highlighted in ***Bold***. Let's get started!

Super Easy Record Restoration Step By Step Guide

If you are new to audio restoration and feel a bit intimidated with the large number of filters and tools in DC EIGHT, then EZ Clean™ is for you. Our first guide is designed to get you going almost instantly. Our EZ-Clean™ filter will remove clicks, pops, hiss and other surface noise from music recording almost instantly. Here we go.

First, we need to run the program. If it's not already up, just ***double click the icon on your desktop*** to start it. EIGHT/DC Forensics starts with a tip screen, which offers quick suggestions as to how to use the program. You can disable this by clicking on the check box in this screen.

In this guide, we're going to use the Classic Source/Destination mode to run you thru the software. You can choose this mode by ***clicking on Edit/Preferences*** in the program. Make sure the check box that is labeled Enable Fast Edit is ***unchecked***. ***If you just unchecked it, close and restart the program*** to get going in Classic mode.

Next, ***open the Demo1 .wav file*** the way we did in the testing section above.

Listen to this file by either clicking the Play button or hitting the spacebar – it's full of clicks, hiss, low frequency noise, etc. It's a mess. ***Stop the playback*** when you're done listening.

And, now for the fun part. ***Click the Filter Menu and choose EZ-Clean***. The filter looks like this:

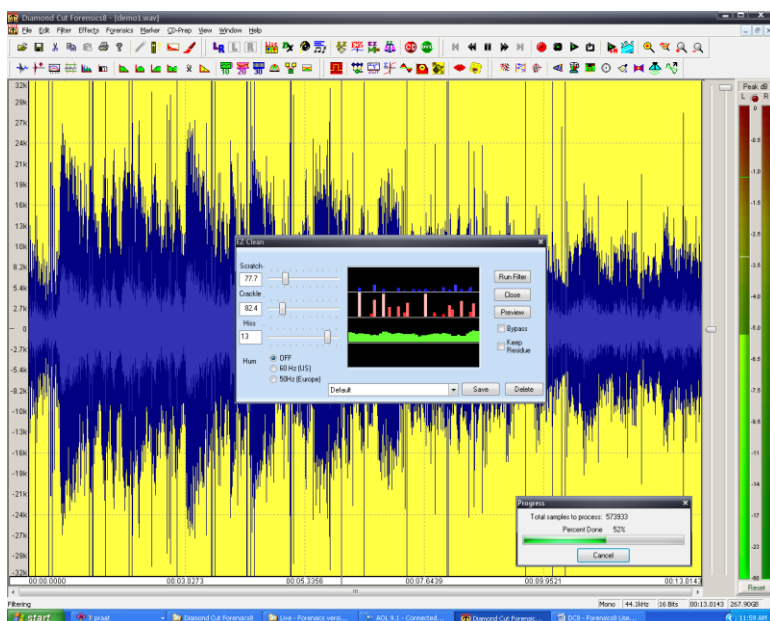


Figure 5 - EZ Clean™ Is Almost Too Easy

Notice that there are only three sliders, one for scratches or clicks, one for crackle (or small clicks) and another for hiss or other continuous type noises. We are simply going to listen to the audio and move these three sliders as we listen.

Move each slider to approximately the settings shown above. We don't need to be perfectly accurate, just set them similar to what you see. When you move these sliders to the left, the filtering becomes more aggressive. *Now click the Preview button.* You'll start to hear the audio.

Listen for a second or two. *Now click the checkbox labeled Bypass.* This "bypasses" the filter and stops the filtering. You are now hearing the original music without the filters in place. Note the large amount of clicks and hiss.

Uncheck the bypass box to start filtering again. What a relief! That's much better isn't it? But, you can do better yet. *Move the Hiss slider*

a bit farther to the left until you get all the noise out. Remember, moving the sliders to the left, makes them filter more, so just *slide them until you are happy with the result*.

Now note the Hum filter on the EZ-Clean screen. If you live in the US, you'll check the 60 Hz box to remove power line hum. If you live in Europe, you'll check the 50 Hz box. There are many other tools to remove larger amounts of hum in DC EIGHT, but this one is quick and easy.

Now, *stop the preview by clicking on the Cancel Button in the Progress box. Click on Run Filter* and you now have a destination file that is fully cleaned! Could this possibly be any easier?

Please continue on with the Easy Restoration guide below. It starts your education on how to use individual tools and goes into more depth on the overall concept of DC EIGHT.

Easy Record Restoration Step By Step Guide

Since EZ-Clean™ is so easy, you don't really get a feel for the overall program. This guide assumes you will use individual tools and will perform a restoration in a series of steps.

You work with EIGHT/DC Forensics by choosing filters to apply to audio. Some of the filters remove noise and others enhance the audio though they are all referred to in this guide as filters. You choose a filter by identifying the type of noise you want to remove and then selecting the filter that removes that type of noise. Make sense so far, right?

In just about every case when clicks and pops are present, we want to remove them first. Trust us, this is the right first step whenever you get clicks and pops on records or other recordings.

To remove the clicks, we'll choose the EZ-Impulse™ filter. Clicks are short noise impulses so it makes sense to use this filter. To choose the filter, *click on the Filter menu item and choose EZ-Impulse Noise*. You'll see the filter open like this:

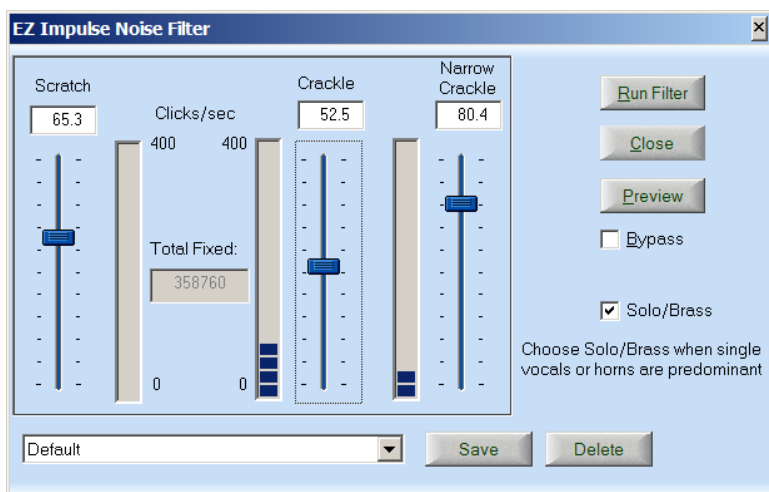


Figure 7 - The EZ-Impulse Noise™ Filter

In the filter window, you will see several features. First, you'll notice sliders that control various filter parameters. These sliders can be adjusted while you listen to the audio so you will instantly hear the result of any changes you make. Our sliders are labeled Scratch, Crackle and Narrow Crackle in this filter.

Next, you'll have Radio buttons that control other aspects of the filters. Again, you can change them while listening, so you'll hear the results instantly. The Radio buttons in the EZ-Impulse filter are Bypass and Solo/Brass.

In every filter, you will have a Preview button. This is the most important button here. This button will start the audio playing while the filter is processing it. You will hear the results of the filter instantly. This makes for easy adjustment of the filters.

Also in every filter, you will have a Bypass checkbox. This takes the filter in and out of the audio stream instantly. When you listen to a filter being applied by the Preview button, you may want to be able to compare the processed audio to the original audio. Clicking this checkbox will bypass the filter and you'll be hearing the original audio. Un-checking it will instantly put the filter back into the signal path.

This way, you can “fine tune” even subtle effects with EIGHT/DC Forensics.

Lastly, every filter will have controls for Presets. A preset is a saved group of settings for this filter. Go ahead and ***drop down the Preset box now*** – it’s the white box at the bottom of the filter. ***Click on some of the presets*** and watch the sliders as they move to good starting points for common tasks. You can tell a lot by looking at the name of the presets. ***Now select the preset labeled Default.*** This one is already set up with good settings for our Demo1.wav file. Every filter will also have a Save and Delete button that allows you to save your own presets under any name you want – and delete them too!

You’re probably ready to try this filter by now, but there is one more thing that is common to each filter that you should know. ***Hit F1 now on your keyboard.*** Notice that our online Help comes up with information on this specific filter. This context sensitive help is available for each and every filter.

Put the Help screen away and let’s clean some audio. If you’ve already been listening to this filter, then shame on you for jumping ahead. Simon has not said click the Preview button yet. As punishment, please go back to page one of this manual and start reading again. We’ll wait for you here.

Welcome back. Now ***let’s get started by clicking the button labeled Preview.*** You will hear the audio as it is being filtered. You will still hear the low frequency rumble and the hiss, but the clicks should be gone. ***Let it preview all the way to the end of the file.*** Notice that once it reaches the end, it will automatically start over at the beginning. This is called Looping and is automatic when you are previewing with a filter (though it can be turned off in the Edit/Preferences screen).

Let’s just confirm that the clicks are gone. To do this, ***check the Bypass box*** in the filter while it is still previewing. Now the filter is bypassed and the clicks will once again be audible. Listen for a while and then ***uncheck the Bypass box.*** Now the filter is again doing its job and the clicks are no longer heard.

It's time to learn how to adjust a filter. While you are previewing, ***move the three sliders to the bottom***. This makes the filter less aggressive and it will filter less. Notice that the clicks return as the sliders are moved down. ***Moving the sliders back up*** results in more of the clicks being removed from the file. All of the filters work this way – you just adjust them while you listen. As you might expect, if you move them too far up you'll make the filter too aggressive and you'll get distorted, stuttering or otherwise bad audio – just move them up enough to get the desired result. ***Set them all to 50***. The “Scratch” control is adjusted by the user to attenuate large impulses. The “Crackle” control is for medium sized crackle and the “Narrow Crackle” control is used to attenuate smaller impulses. “Solo/Brass” should be turned on when dealing with up-front vocals or brass instruments like horns.

Now ***click on Cancel*** in the Progress window seen here:

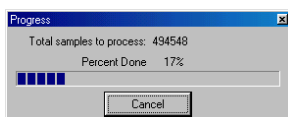


Figure 8 - Monitor your progress

This stops the preview from playing. By Previewing, we have adjusted the filter and confirmed that it is doing its job. The next step is to ***click on the Run Filter*** button. This takes the filter just as you have set it and applies it to the demo1.wav file and creates a new file in the lower window. This new file has been run through the filter and has the clicks removed. At this point, ***click on Close in the EZ-Impulse filter*** window since we're now done with it.

Look at the two waveforms. The top one is called the Source. This is where we normally work on a file, preview filters, etc. The bottom is called the Destination and is the result of our filtering efforts. You can play either one by clicking in the respective window. You have not changed your original file at all – rather we've created a new cleaned up version.

Now it's time to remove that rumble sound, but how can we do that if we work on the Source window and we really want to remove the rumble from the semi-clean file in the Destination window? The

answer is a little command under the File menu called Make Destination the Source. This moves the file in the bottom window up to the top where we can work on it. ***Click on File/Make Destination the Source now.***

A “Save As” box will come up and suggest a new name for this file; just ***click on Save*** to accept it.

Note: EIGHT/DC Forensics will automatically assign sequential names to new files. While you’re new to the program, always just accept its recommendation as to file names.

Now we are ready to remove that low frequency rumble. To do that, we’ll choose the High Pass filter. ***Click on Filter/High Pass now.*** (To get there, go to the Filter Menu\LP BP HP Filters and then select the “High Pass” filter.) A high pass filter will remove all frequencies below a certain value and allow all higher frequencies to pass. Drop the Preset box and ***select the preset called “Demo audio Wave file de-rumble”***. It’ll look like this:

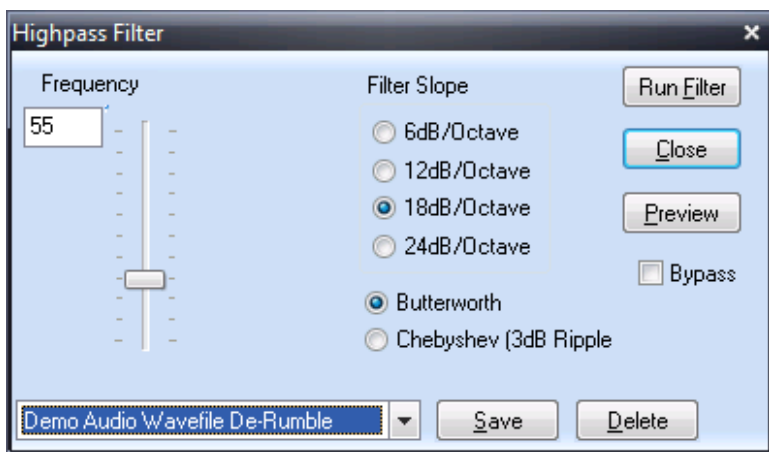


Figure 9 - The Highpass Filter

We’ll start to go a bit quicker now since you already know what most of the buttons and controls do on this filter. To find out specific info on this filter, don’t forget you can call for help by pressing the F1 key. We are going to free you now to preview and play with this filter on your

own, but when you're done, return it to these settings by once again clicking on the demo Wave file preset. **Run the filter** when you are ready.

You now have removed two of the annoying noise types in this file. First, we rid ourselves of the clicks and pops, and then we removed the rumble. Now it's time to get rid of that loud hiss sound. First, remember to **move our Destination file up to the Source window**.

Now **click on Filter and select the Continuous Noise Filter**. This filter is perfect for hiss and other types of continuous noise. Also, **click on View and make sure you have Time Display checked at this time**. The Time Display box shows you various timing calculations with the program and will come in handy, as you will see. Your screen should look something like this:

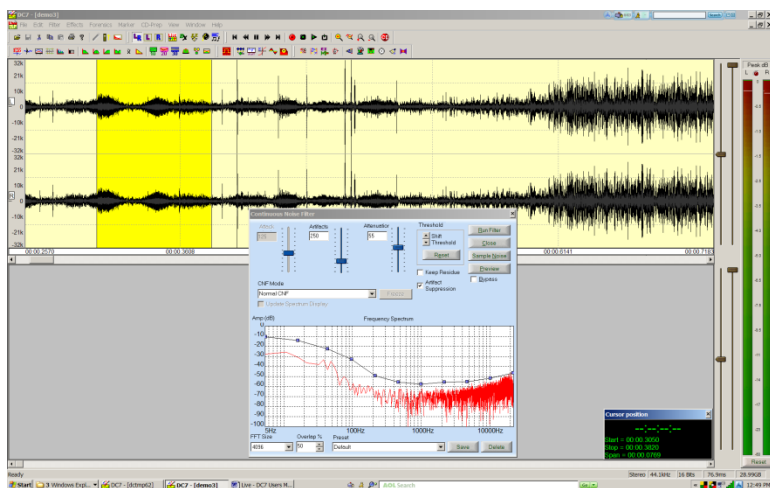


Figure 10 - The Continuous Noise Filter is Awake

The Continuous Noise filter is one of the coolest in the whole program so follow along carefully here. This filter will remove just about any continuous noise in a program, but it needs you to give it a sample of this noise. Once it is able to examine the noise, it will be able to seek it out and get rid of it. **Play the file now using the Play button at the top** and listen carefully to the first couple of seconds. You will hear an area

right at the beginning, which contains only the noise and no good audio. That's a great spot from which to grab a sample of the noise.

To do this, we need to click and drag with our mouse to select this area. The area highlighted in yellow in the illustration above is the area we want to select. Just left click about ½ inch from the left edge and, while holding down the left mouse button, drag all the way left. Let the mouse button go and you will see an area highlighted in bright yellow. This is the area you have selected. Move the mouse pointer over either edge of the selected area and the pointer will turn into a Left/Right indicator. ***Click and drag as necessary to select an area from the beginning of the file that is about .5 seconds long. Use the "Span=" display in the timer window to confirm that your selection is around ½ second long.***

Now hit the spacebar. You'll play only the selected area. It'll be quick. This allows you to audition a selected area to make sure you are really working on the correct area (does not contain silence or desired signal). You should only hear the hissing noise. Again, your selected area should look like the one shown above.

It's time for the fun part. You will like this. ***Hit the Sample Noise button in the Filter window.*** The filter will analyze the noise sample and will display the frequency characteristics of the noise in red. The blue line is the filter that has been designed to attenuate this noise. Notice how they track with each other. Yours should look something like this:

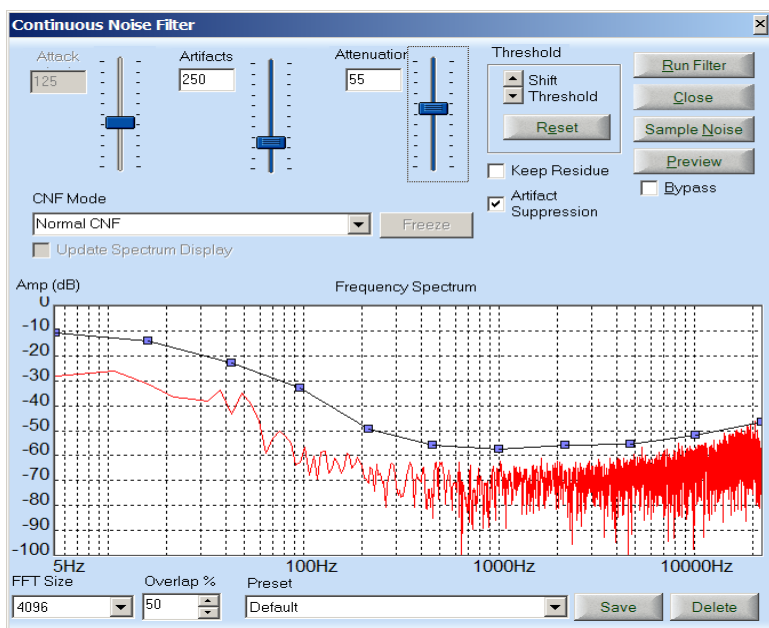


Figure 11 –The Continuous Noise Filter Does its Job

If you have jumped ahead again and clicked on Preview, there will be no lunch for you today. First, we've got to explain those little blue dots on the filter line. Notice how the blue line is above the red noise sample? The higher the blue filter line, the more filtering occurs at that frequency. The blue dots are called Control Points and allow you to adjust the amount of filtering for frequency components in the vicinity of the control point manually. We're going to do that in a second, but realize that the Help file contains the complete information on this filter.

Before we click on Preview, we need to again select the whole file so we'll hear the whole thing and not just our ½ second noise sample that is now selected. To select the entire file, just ***double click anywhere in the waveform display area***. Notice how the entire thing gets highlighted in yellow. ***Now, go ahead and click on Preview.***

Listen to the audio. The clicks are gone. The rumble is gone and the hiss is gone! Well, not quite. You can still hear a bit of hiss can't you?

While Previewing, click the bypass button a few times to take the filter in and out of the signal path. Yes, the hiss is reduced, but there is still some of it there. Let's adjust this filter to get rid of all of the hiss.

Make sure your bypass button is NOT checked and make sure you are previewing the audio. Now grab the 2nd control point from the right side and move it up a bit as shown below.

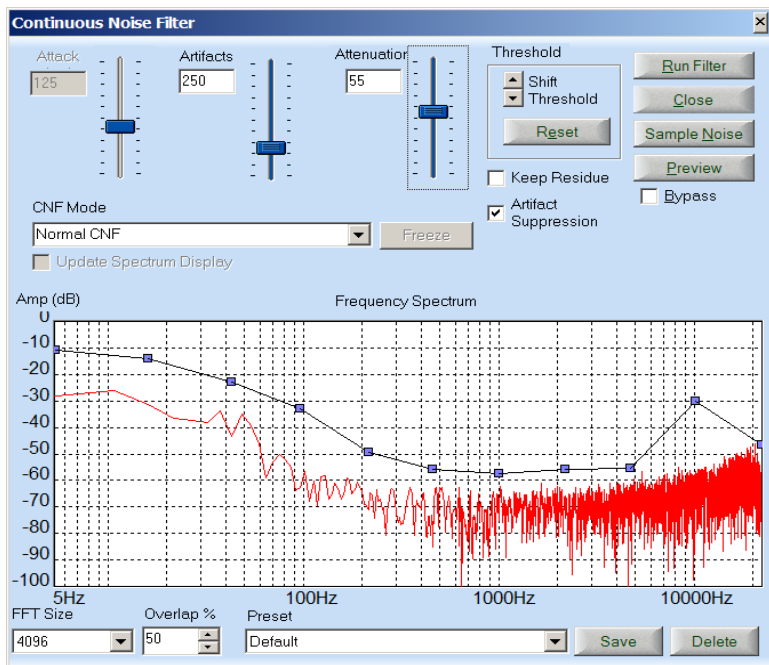


Figure 12 - Tackling the remaining Hiss

It's like magic, isn't it? The hiss completely goes away because you told the filter to be a bit more aggressive on the high frequencies. Remember, moving the blue line up makes it filter more. Now *stop the preview, click on Run filter, and close the Continuous Noise filter.*

You now have a Destination file that is completely restored. To finish, *click on File/Make Destination the Source, accept the file name and you're done.* This file is the completed version and all others can be deleted or saved if you want each processing step before and after the

saved file. You can now exit the program or close the files using the commands under the File Menu.

What you've learned in this section

In this quick Step by Step guide, you've learned how to launch the program, how to select filters, how to preview them, how to bypass them, how to load and save presets, what are Source and Destination files, how to apply multiple filters to a single file, how various filters work and much more. Not bad for a few minutes! Next, we'll give you a more advanced Step by Step in which you get to use the most powerful feature of EIGHT/DC Forensics, the Multi-Filter.

Advanced Record Restoration Step By Step Guide

You learned a lot in the last Step-by-Step Guide, so let's continue by restoring this same file using a completely different approach. This will show you the flexibility and power of EIGHT/DC Forensics and will introduce you to our powerful Multi-Filter. Also, we will do this advanced restoration Step-by-Step Guide using our new Fast Edit Single Screen mode. We'll assume you've done the Basic Guide above and are therefore comfortable with the basic functions of the program so we will take this at a little faster pace. As before, the actions you are expected to perform will be in **Bold** type. Let's get started!

First, we want to make sure that you are in Fast Edit mode. To do this, ***click on Edit/Preferences and make sure the box labeled Enable Fast Edit is checked.*** If it's not already checked, ***put a check mark in the box by clicking it.*** Now ***close the program and then restart it.*** The operational mode change will only take effect when you start the program.

Remember, Fast Edit mode is a single screen mode of operation with no Source or Destination file window. This mode of operation has the advantage of allowing almost instant cuts, pastes and copies on even very large files. It also may be more familiar to users of traditional audio editing programs. You can change from Fast Edit to Classic Mode anytime you want.

Next, we want to ***enable the Fast Edit History box by clicking on View and then checking Fast Edit History.*** This will open your History

box. This box will show you everything that you do with this file in the order you do it. It also allows you to return to an earlier state of the file quickly and easily.

Now open the *Demo1.wav* file in the same way we did before. **Select an area near the center of the file using a mouse click and drag.** Your screen should now look similar to this:

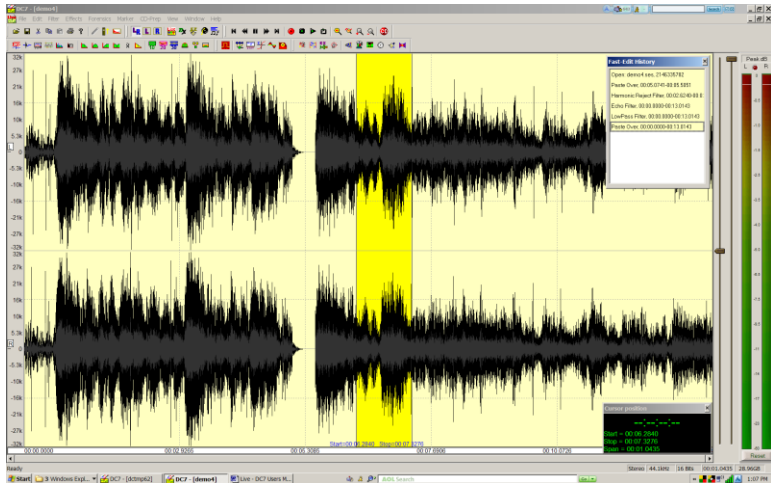


Figure 13 - A Highlighted Section in Fast-Edit Mode

Let's cut out the highlighted area. **Just click on *Edit* and then *Cut*.** Boom! It's gone! Fast Edit mode is called that because it's *FAST*. You will now have two entries in your History window – the top one is for the original file as opened, and the second one for the version you now see in front of you which is somewhat shorter.

Click on the first entry in the Fast-Edit History window now. The original file instantly comes back! Did I say this thing was fast? Everything you do will show up in the history window and you can bounce around your various versions of a file at will. **Click on the first and then second entries a few times** to confirm this and then **right click on the second entry and choose "Delete"**. You just deleted the cut-up version and we're back to the original file.

Now ***click on Filter and choose Multi-Filter. Double click anywhere in the waveform display*** to select the entire file for processing or left click and drag to select a segment of the file. Notice how EIGHT/DC Forensics allows you to select an area or the entire file while the filter windows are open. This is a very handy feature.

The Multi-Filter is one of the most powerful features of EIGHT/DC FORENSICS. It allows you to string together several filters and preview or apply them sequentially or all at once. This saves a lot of time and makes it possible for users to create some incredibly sophisticated filter setups.

Notice that the Multi-Filter has an input on the left side and an output on the right side. Between the input and output is a signal path. You can drag any filter you want into this signal path. At this time, ***remove any filters that are in the signal path by clicking on them and dragging them out of the signal path.***

Remember when we did the Basic Step by Step Guide (above), we used three filters: the EZ-Impulse™ Noise filter for clicks, the High Pass filter for rumble and the Continuous Noise filter for the hiss. ***Drag these three filters into the signal path*** with the EZ Impulse filter on the left side followed by the High Pass filter in the middle and then the Continuous Noise filter on the rightmost side. Now ***double click on the EZ-Impulse filter icon*** in the signal path of the Multi-Filter. Your screen should look something like this:

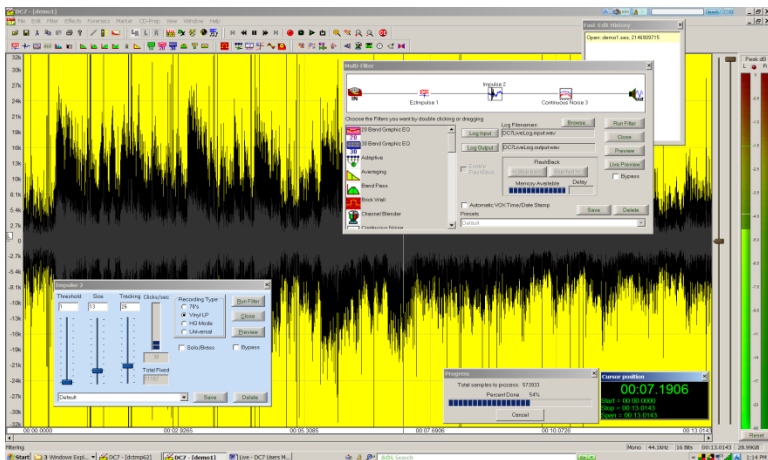


Figure 14 - Working with the Multi-Filter

To open the individual settings for any filter in the Multi-Filter signal path, just double click on that filter and make your settings. For the Expert Impulse filter, ***drop the preset box and choose “Demo Audio Wave file De-click”***. You could preview and adjust while listening here, of course, but we’ve already got some good settings available as a preset. ***Close the Expert Impulse filter window.***

Now ***Double click on the High pass filter and select the “Demo Audio Wave file De-Rumble” preset.*** ***Close the High Pass filter window when done.***

Lastly, ***open the Continuous Noise filter and choose the “Demo Audio Wave file DeNoise” preset.*** ***Close the window.*** We used these presets only because they are available. You can always choose any preset or you can set the filters yourself by just previewing as we did in the Basic Step by Step guide.

Note: Each individual filter can be previewed even in the Multi-Filter.

Now we have three filters strung together. ***Click the Preview button in the Multi-Filter window.*** You will hear the audio being filtered by all three of these filters. ***Click the Bypass button a few times*** to marvel at

what is happening here. Feel free to *giggle* and then *cancel the preview*.

Now we're going to add a fourth filter. **Drag the Paragraphic EQ icon into the signal path** and put it last in line (on the rightmost side). **Double click the Paragraphic EQ icon to open its adjustment window.** Now again click on the Multi-Filter Preview button. You may have to move the Paragraphic filter window out of the way to access this button. You are now hearing the audio filtered thru 4 filters. **Make sure the Bypass button is not checked.** While the preview is happening, find the blue control point in the Paragraphic EQ that is close to 110 Hz. You can see this on the horizontal scale on the bottom of the EQ adjustment area. **Move this up** to increase the bass frequencies in our old recording. The more you move it up, the more low frequencies you get. It should look something like this:

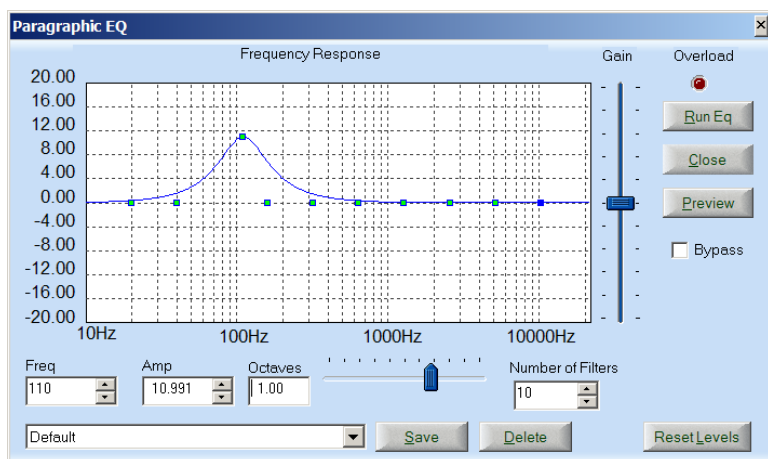


Figure 15 - Using the Paragraphic EQ

You should notice that you are applying all four filters at once and you are adjusting one of them. The results of your adjustments are instantly heard while all the filters are being applied. **Cancel the preview, run the filter, and "Save Source As" the file.** Fast Edit mode requires that you save files when you are done.

What you've learned in this Advanced Step-by-Step Guide - - -

You now know how to choose and use Fast Edit Mode, how to bring up and use the Multi-Filter, and how to Preview and adjust while using the Multi-Filter.

Forensic Audio Step-by-Step Guide (DC FORENSICS Only)

If you are a new user of DC FORENSICS, please go through the Basic and Advanced record restoration guides. We know you won't be restoring records with DC FORENSICS, but these guides provide an excellent introduction to the overall usage of the program. We'll assume you have this basic knowledge for this Forensic Guide. As before, the specific steps for you to perform are in ***Bold***.

To get started, ***run DC FORENSICS and put it in Classic (Source/Destination) mode. Now open the file*** called Forensicsdemo.wav. Forensics version demo files can be found on the start menu.

Play this file. Like a lot of Forensic files, the signal to noise ratio is very bad. The noise is much louder than the voice and this makes it difficult to understand what is being said. The speech is masked by a loud buzz and has a high frequency and a low frequency noise throughout the piece. As always we ***start by listening so that we can identify the noise components that make up our problem.***

It's easy to hear in this file that we will need multiple filters to clean this thing up. Therefore, let's use the Multi-Filter. ***Click on Filter/Multi-Filter*** and bring it up and ***clear out any filters that are in the signal path.*** The file contains a loud Buzz that almost overwhelms the good audio. To remove hum and buzz, we will use the Harmonic Reject Filter. It's called this since hum and buzz are rich in harmonics and this filter can remove them. We'll need one of these filters for sure but don't put it in the Multi-Filter yet.

Next, ***listen again.*** There are other noises in this piece that you probably can't easily identify by listening. However, we don't only have to listen to the audio, we can also see it. ***Click on View and then choose the Spectrum Analyzer. Set the Spectrum Analyzer to a***

resolution of 43.07 Hz and 1024 FFT size. Now **click on Play** on the toolbar and you'll see the audio as well as hear it. After listening for a few seconds, **hit the spacebar to stop the playback.** The Spectrum Analyzer continues to show you the frequency spectrum of the audio. Notice that there are some “spikes” of audio in here. One, for example, is at around 950 Hz. You can click on any spot in the display and the Spectrum Analyzer will show you the frequency and amplitude at that spot. It should look a lot like this:

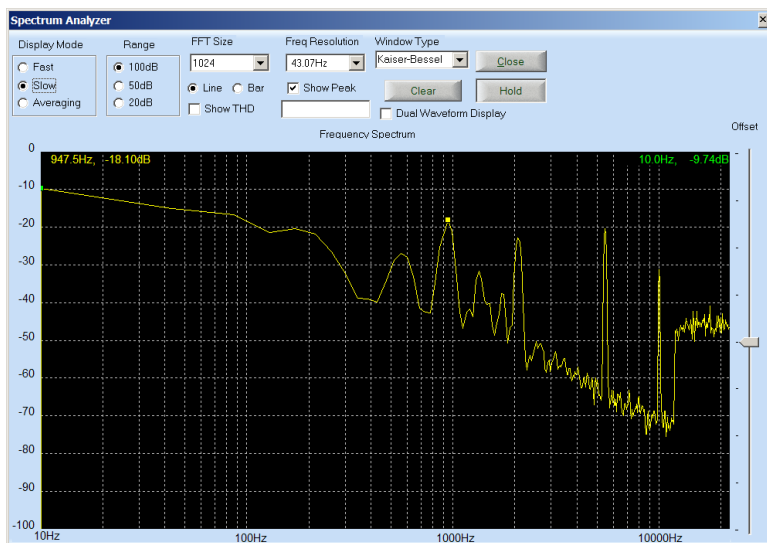


Figure 16 - Using the Spectrum Analyzer

If you look closely, you can see that there are actually four spikes and some random noise above 12,000 Hz. We can remove specific frequency noise with a Notch filter and we can remove that high frequency noise with our Continuous Noise filter or a Low Pass Filter. So now we know all the basics to attack this problem – we'll need a Harmonic Reject filter, 4 Notch filters and a Continuous Noise filter. You can see how we used the tools to come up with a general plan of attack. Now, **close the Spectrum Analyzer.**

We have already created a Multi-Filter preset designed to clean this file. **Select the Multi-Filter preset called “Forensic Demo Clean Up Filter”.** Notice that our basic filter is not quite the same as we had set-

filters has its own tutorial in Section Two of this Manual). ***Close each filter*** when you finish looking at the settings.

Brick Wall Filter – this is set to reject all low frequency sound below 135 Hz and all sound above 11 kHz. Remember the noise in the Spectrum Analyzer at 12 kHz and above? We took it out with this filter.

Harmonic Reject Filters – both are set at 60 Hz, which is a common power line induced noise frequency (50 Hz in many countries outside the US). Why do we use two of them? Because multiple filters of this type will deepen the attenuation of hum and buzz noise. It's like one worker digging a hole two feet deep and then another also digging a hole two feet deep, but the second worker starts in the hole created by the first. You end up with approximately a four foot deep hole.

Spectral Filter – Remember we thought we might need 4 notch filters? We could have done that, but our Spectral Filter allows us to set up as many as 32,000 notch filters at once! Additionally, this filter allows us to either boost or cut any frequency range we want including ones like these that are not related to each other harmonically. This is just a more efficient way to handle several unrelated noises.

Continuous Noise Filter – This filter is set up to aggressively filter just about all frequencies outside the speech range. Remember, this doesn't take out all sound, but targets noise while leaving behind good audio. It's a good filter to add to the mix.

Median Filter – This was chosen to add intelligibility to the voice. It is good for enhancing muffled speech.

Virtual Valve Amplifier - This will add harmonics to the voice, increasing intelligibility.

Paragraphic EQ – This is set to raise the volume of the speech frequencies.

So you see, we really did follow our initial path by using Harmonic filters, Notch filters and Continuous Noise filters. We ended up adding some additional tools that further enhanced our audio. You will likely

do the same thing as you perform your own restorations. This process of first planning a course of action and then making it better is a common one that you will use often.

Close all filters, Make Destination the Source (saving it under a new name), and exit the program.

Live Mode Step by Step Guide

This guide will introduce you to another major feature of the program. As before, follow along with the included demo file and perform the actions printed in **Bold** type. We assume you have mastered the other concepts we introduced in the Basic and Advanced record restoration Step by Step guides. Please make sure you have completed these earlier guides before attempting this one.

Live mode is a powerful feature of EIGHT/DC FORENSICS that allows you to remove noise and enhance audio as it happens. You do not have to record the audio to the hard disk before applying filters. This is useful for cleaning up radio reception for Hams, SWLs, DXers, and Forensic users. If you have your computer connected to your home stereo, you can use Live mode to actually restore your records while you play them. It can also be used “on site” by Forensic users who may need to listen to (and record) surveillance audio as it happens.

To use Live mode, ***make sure you have your speakers connected to the sound card output and an audio signal connected to the line level input.*** This audio signal can be from a radio, turntable (with preamp) or any other signal. For this Step-by-Step guide, we’re going to assume you are using a radio to provide a signal to the program. In this case, you would ***connect the headphone or other audio Output of your radio to the Line Input of your sound card.***

Turn on the radio and you should hear audio from the speakers connected to your computer. This step depends somewhat on the brand and model of sound card you have, but the great majority of sound cards will simply take whatever audio appears on the input and send it directly to the output of the sound card. If you don’t hear anything, just keep going in this Guide, as we’ll check everything out a bit later.

Now **turn off your Radio** and **run the program** if you haven't already and then **open the file called "Radiodemo.wav"**. It's in the same directory as the other demo files. Before we do this live, we're going to experiment a bit with a recorded bit of audio.

Once the file is loaded, **listen to it**. You can hear that there is A LOT of noise in this file. It is understandable, but would not be fun to listen to due to all the noise. Once the file has finished playing, **open the Multi-Filter** (under the Filter Menu) **and choose the preset called "Live Mode Demo Cleanup – SW Radio"**. This preset provides some useful tools for this demonstration file. This preset will also be a good starting point for your own efforts as you use the program in Live mode.

Double click the Continuous Noise Filter to open its adjustment window. You can look at the settings for all the tools in the Multi-Filter preset and you should be able to understand what they are doing based on your learning so far. Remember, you can always open a specific filter and then hit F1 to find out about it. Your screen should look something like this:

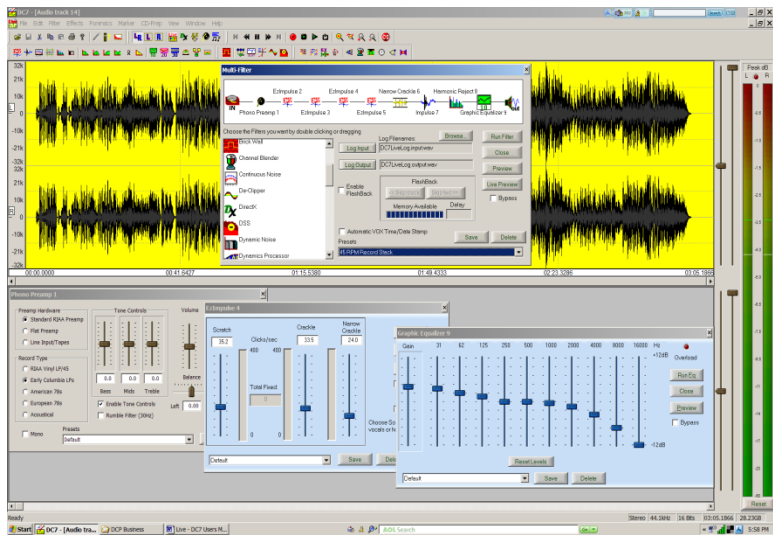


Figure 18 - Multiple Screens Shown Working Together

We want to bring your attention specifically to the Continuous Noise filter here because we are dealing with a concept that you might not have considered – How do I use this filter in Live mode when I can't take a sample of the noise? Remember, the audio is happening in real time and is typically not recorded to the disk at all in Live mode. It'll therefore be impossible under these conditions to actually take a noise sample. Our preset setting here shows how you can deal with this. Notice that the filter is set to provide a fairly large amount of attenuation to the low frequencies and the higher frequencies (remember, the higher the blue square controls are set, the more noise is removed at that frequency). With radio signals, we will seldom care about very low or very high frequencies anyway, so this setting gets rid of a lot of noise without unduly bothering our good signal. The controls between 300 Hz and 3,000 Hz (which are speech frequencies) were then simply adjusted for good results. Go ahead and ***hit the Multi-Filter Preview button now. Click Bypass in and out*** to hear the improvement.

Notice the slider labeled Attenuation. This control determines the overall amount of filtering that is going on for the entire filter. If you move this control upwards, it will filter more overall noise. ***While Previewing, adjust the attenuation slider to find the point you like best.*** You'll find that this very weak signal is now quite understandable – if you speak German. The other filters are also adjustable, of course. You'll quickly find that there is a tradeoff here. The more aggressive you make the Continuous Noise filter, the more digital artifacts you will induce. Just adjust it to please your ears.

Now ***stop the Preview*** and let's go into Live mode. ***Turn on your radio*** so that you can hear the signal in your computer speakers.

Note: if you don't hear anything continue on anyway as some sound cards don't monitor incoming audio by default. ***Double click the leftmost icon in the Multi-Filter*** so that you can adjust your audio settings. Do that and then ***click on the button labeled "Live Preview"*** and the program will immediately start providing filtered audio to your speakers. You should certainly be hearing audio now.

In the Live mode, your screen should look something like Figure 19.

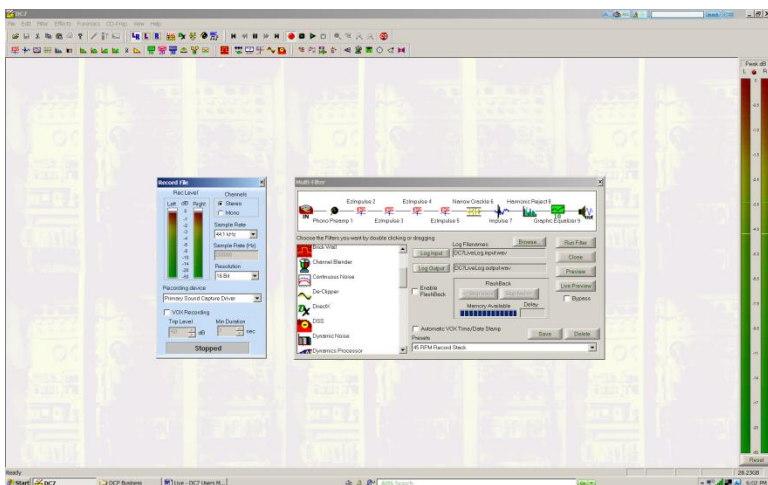


Figure 19 - Live Mode is Engaged

Important -- Do you hear two versions of the audio – one a bit later in time than the other - like an echo effect? If so, you need to turn down the input of your sound card. If you have a standard sound card, just double click the speaker icon in your tray and choose Options then Properties. Now click on Recording and then click “OK”. Now move the slider labeled Line Input or Line down to the bottom. The echo should go away! If you have a more sophisticated sound card, refer to its documentation to find out how to turn off monitoring or turn down the Line Input control.

If you have the DC Forensics version of the program, you cannot only process audio in real time, but you can record it as well. Just click the Log to Disk button. Additionally, you can also select VOX recording (the program starts and stops recording automatically with the signal that comes in) and have the program mark the segments with the date and time of their occurrence.

Here’s a tip. You can now add or change the filters as you please. A good one for radio work is the Dynamic Processor using the ALC mode. This can help a lot with fades in a signal as it will even out the audio in real time.

What you have learned:

In this Step-by-Step Guide, you've learned how to use the Multi-Filter in Live mode to process audio as it happens. You've learned how to adjust your sound card for Live mode – remember, however, that some low quality sound cards cannot record and play at the same time and therefore can't be used in Live mode. You also learned how some filters could be used to make radio reception much better.

Which Tool Do I Use?

Probably the most frequently asked question we get each day is “How do I get rid of a certain type of noise?” Certainly, the more you use this product, the more proficient you'll become at picking the right tool to match the noise you're trying to remove. But for now, if you're new to the game, we've included this handy chart that may help you get a jump on finding the tools that match your problem areas.

Filter Finder

<u>Sound Restoration Category</u>	<u>Sound Defect or Task to be Performed</u>	<u>Filter Type</u>
Early Acoustical Cylinders and Discs	"Pops"	Impulse Noise (EZ-Impulse™ or Expert Impulse Filter)
	"Crackle"	Average, Median or Continuous Noise Filter
	"Distortion"	Low Pass Filter or De-Esser in the Dynamics Processor
	"Hiss"	Dynamic Noise or Continuous Noise Filter
	"Rumble"	Highpass or Continuous Noise Filter
	"Thin Sound"	Graphic Equalizer
	"Reverse Skip"	Cut
	"Forward Skip"	Copy and Paste Insert
	Skip / Miss-tracking	Speed Change Filter / Fractional Speed Mastering
	Cracked Record	Big Click Filter

	"Thumps"	Selective use of the High Pass Filter (set to 6 dB/Octave)
	"Off Pitch"	Speed Change Filter
LPs & 45 RPM Records	"Ticks"	Impulse Noise or EZ Impulse™
	"Pops"	Impulse Noise or EZ Impulse™
	"Distortion"	Low Pass Filter
	"Rumble"	High Pass Filter
	"Shrill"	Graphic Equalizer
	"Reverse Skip"	Cut
	"Forward Skip"	Copy & Paste Insert
	Noise between Cuts	Dynamics Processor – Expander / Gate
	Muddy Bass on Stereo Records	Channel Blender
	Stereo "Ping-pong" effect	Channel Blender
	"Off Pitch"	Speed Change Filter
Magnetic Tape Recording	"Hiss"	Dynamic Noise Filter or Continuous Noise Filter
	"Highs Loss"	Time Offset (azimuth correction)
	"Smeared" Stereo Image	Time Offset (azimuth correction)
	Clipping Distortion	De-Clipper
	"Off Pitch"	Speed Change Filter
	"Hiss"	Dynamic Noise Filter or Continuous Noise Filter
AM Radio or Short Wave Radio		
	"Static"	Impulse Noise (EZ-Impulse™ or Expert Impulse Filter)
	Line Frequency "Buzz"	Harmonic Reject Filter
	Volume Fading	Dynamics Processor ALC
	Heterodyning "Whistle"	Notch Filter or Multiple Notch Filters
AM Broadcast	"Whistle"	Notch Filter (Europe - 9 kHz) (US - 10 kHz)
	Line Frequency Buzz	Harmonic Reject Filter
	Volume Fading	Dynamics Processor ALC
FM Stereo Broadcast		
	Multi-path Distortion	Dynamics Processor / De-Esser or Channel Blender
	Multiplex Noise	Channel Blender
	Ignition Noise / Static	Impulse Noise Filter
	"Feedback"	Notch Filter
	"Hum"	Notch Filter
	"Mic 'P' Pop"	Highpass Filter
	"Dead"	Dynamic Noise Filter (spectral enhancement mode)
	"Digital Sound"	Virtual Valve Amplifier / Tube Warmth

Telephone Conversation	Street Noise	CNF Spectral Subtraction
	HVAC Noise	CNF Spectral Subtraction
	Hum	Notch Filter
	Buzz	Harmonic Reject Filter
	Sibilant "Ess"	De-Esser
	Digital Clipping	Limiter Preset in the Dynamics Processor
	"Intelligibility"	Bandpass Filter
	"Noisy - Random Out of Band"	Brick Wall Filter in Bandpass mode or Band Pass Filter in Chebyshev Mode
	"Noisy – In Band"	Continuous Noise Filter
	"Muffled or Garbled"	Median Filter or Spectral Inverse Filter
	Variation in loudness between parties (near party/ far party gain compensation)	Dynamics processor / Compressor / ALC
		Punch and Crunch in ALC mode
	Cell Phone Noise Interference	Cell Phone Noise Filter
	Cancellation of Radio / or TV using a reference track	File Conversions (Left – Right Mode)
		Adaptive Filter in Reference Mode
Surveillance Recording	Automatic Voice Activated Recording	Continuous Filter in DSS Mode
	Automatic Time Activated Recording	Multi-Filter operating in VOX mode
	Noise isolation and identification	Timer Recording Function
	Poor Speech Articulation	Slot Filter
		Median Filter w/ Weighting
	Varying Random Noise Profile	Adaptive Filter
	Varying Coherent Noise Profile like music playing behind speech	Continuous Noise Filter in DSS Mode with reference track
	DTMF Tone Amplification	DTMF Presets found in the Paragraphic EQ
	DTMF Tone Identification	Spectrum Analyzer
	Voice Masking	Speed Change + Stretch & Squish Filters
Forensics Audio Analysis & Enhancement	Voice Disguising	Voice Garbler
	Gunshot Ballistic Timing	Markers and Time Display / Spectrum Analyzer
	Gunshot Ballistic Identification	Time Display and Spectrogram
	Matching	
	Voice Identification	Spectrogram or Voice ID

	Analog Tape Deck Azimuth Correction	Time Offset Adjustment in the File Converter
	Person Talking too fast for Transcription Poor Intelligibility	Stretch & Squish
	Voice Pitch too High Tape Authenticity	EZ Forensics Spectral (Inverse) Filter Speed Change Filter Spectrum Analyzer
	Reference Channel Acoustical Compensation Poor Intelligibility Adaptive Noise Rejection	Bandpass Filter plus 20 Band Equalizer plus Reverb 20 Band Graphic EQ AFDF in the Continuous Noise Filter Adaptive Filter under Forensics Menu (TDAF) 30 Band EQ Spectral Inverse Filter or Overtone Synthesizer
	Narrowband Noise Rejection De-Muffling	
Optical Movie Soundtracks	"Pops"	Impulse Noise
	"Crackle"	Median or Crackle Filters
	"Thuds"	Highpass Filter
	"Hollow"	Graphic Equalizer
	"Film Flicker"	Harmonic Reject Filter
Television / Video	Vertical Sync Pulse Bleed-through buzz	Harmonic Reject Filter – 30 Hz United States 25 Hz Europe
	Horizontal Sync Pulse Bleed-through	Low Pass Filter
Any Sound Source	Mike "P" Pop	Highpass Filter selectively applied
	Low Gain or Volume	Gain Change (under Edit menu)
	Acoustical Feedback	Notch or Harmonic Reject Filter
	Clipping Distortion	De-Clipping Filter or Manual De-Clipping Process or Lowpass Filter selectively applied or Impulse Filter
	De-Ess (excessive sibilance of the pronunciation of the letter "S.") Pitch incorrect	Lowpass Filter selective applied, or use the Dynamic Processor/ De-Esser Change Speed Filter

Line Frequency “Buzz”	Harmonic Reject Filter or Narrow Crackle Filter
Excessive Dynamic Range	Dynamics Processor or Punch & Crunch
Lacking Dynamic Range	Dynamics Processor or Punch & Crunch
Top Octave missing	Virtual Valve Amplifier / Harmonic Exciter
Top Octave Weak	Dynamic Noise Filter Enhancer Mode
Bottom Octaves Missing	Sub-harmonic Synthesizer
Recording lacks “warmth”	Virtual Valve Amplifier
Top Octave Missing	Overtone Synthesizer
Too much Reverb	Continuous Noise filter
Weak Vocal	Gain Change selectively applied
No Ambience	Reverb
Digital Clipping	De-Clipper
Analog Clipping	De-Clipper
Harmonic Distortion	Polynomial Filter / De-Esser
Inter-modulation Distortion	Polynomial Filter
Multi-Tone Whistle	Spectral Filter or Multiple Notch Filters
Signal Over modulates	Limiter Presets in Dynamic Processor
Noise Between Tracks or Rides	Noise Gate Presets in Dynamics Processor
Automatic Noise Reduction	EZ Clean™ Filter
Sterile Bass Sound	VVA Fat Bass
Medium Resolution Frequency Contouring	20-Band Graphic EQ
High Resolution Frequency Contouring	30-Band Graphic EQ
Normalize Loudness Between Multiple Wave files	Auto Leveling in the Batch File Processor
EQ Matching	Spectral Difference Filter
Occasional Spurious Noises	Interpolate or Spectral Interpolation

Section 2 – The System and Its Operation

In this section we'll discuss everything you'll see when you open EIGHT/DC FORENSICS for the first time. These items are arranged in their Menu order...meaning that we start at the left, open the first menu and proceed from there. We'll also discuss the various ways to access these menu items. In EIGHT/DC FORENSICS, many items in the Menu bar can be accessed via Hot Keys and icons on the Tool Bar.

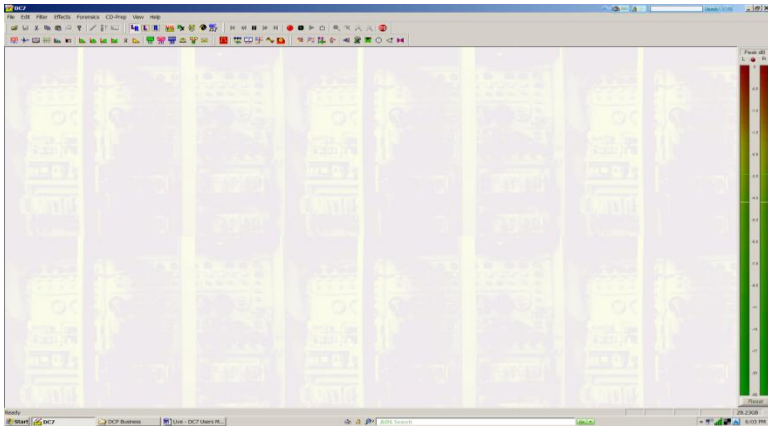


Figure 20 - EIGHT/DC FORENSICS Before the Source File is Opened

Important Note: If you see a Tool or a Menu Item listed here but don't see it in the program, keep in mind that many items are grayed out or not available until an audio file has either been recorded or opened within the program (.wav, mp3, wma, etc).

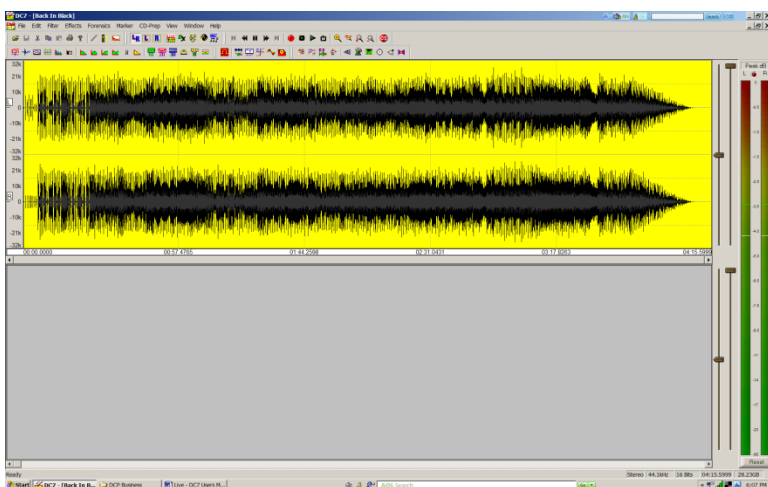


Figure 21 - Menus Awaken after an Audio file is opened

The File Menu



Open Source

Data in the following file formats can be opened:

- Wave Files (*.wav) †
- BWF (Broadcast .wav)
- Vorbis (Ogg Vorbis) (Lossy Compression) (.ogg, .oga)
- WMA Compressed Audio (.wma) {does not support files encoded with DRM (Digital Rights Management)}
- MP3 Compressed Files (*.mp3)
- FLAC Files (*.flac) (Non-Lossy Compression)
- AIFF Files (*.aif, *.aiff)
- Video Files (*.avi, *.asf, *.mpg, *.mpeg)
- All Files (*.*)

† The following compressed Wave File formats are also supported:

- A-Law

- Mu-Law
- ADPCM

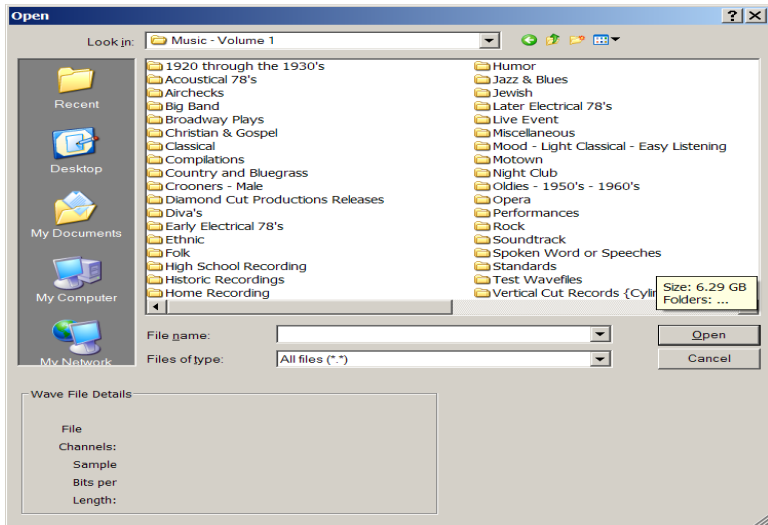


Figure 22 - Open Dialog Box

This command opens the desired audio file on which you will perform the EIGHT/DC FORENSICS processes. The file will be displayed as a periodic sampling of the file's peak amplitude envelope vs. time in the Source graphical workspace. Please note that the software allows you to load .wav, MP3, .wma, .aif and .aiff files as well as A-Law and Mu-Law compressed files and automatically converts them to .wav files upon opening. If you click once on a prospective file, the Wave File details will be displayed (File Size, Channels, Sample Rate, Bits per sample, Time Length of File). This function can also be activated by using the file folder icon in the File Tool bar in the upper left hand corner of the workspace or by using the **Ctrl + O** hot key.

Variable Bit Rate (VBR) MP3 File Support

MP3 files of Constant Bit Rate (CBR) and Variable Bit Rate (VBR) are supported automatically within EIGHT/DC FORENSICS. They simply open with no additional input from the user.

Convert Various Formats to .wav (Tutorial)

EIGHT/DC FORENSICS provides you with a means for converting various file formats into .wav extension files for editing. You can perform your signal processing after the conversion to .wav and then re-convert back to your favorite format with an appropriate encoder if necessary. The MP3 decoder supports Blade, Lame, Fraunhofer, and many other formats. Also, it can handle formats with or without an ID3 tag and with either fixed or variable bit rates. The process of converting from an MP3 (*.mp3), AIFF or A-Law and Mu-Law to wave (*.wav) is extremely simple:

1. Click on the File menu with the left mouse button
2. Click on the “Open Source”
3. Under the “Files of Type” selector box, find your file type and click on it
4. Use the “Look In” box to find your file, select it and Click on “Open”
5. The file conversion will begin
6. After a period of time, a waveform will appear in the Source Workspace. This is a 16 bit, converted .wav file representation of the compressed file. The original file determines the sample rate and number of channels.
7. It will have the same name as the original file except it will have the .wav extension
8. If there is already an existing .wav with the same name, a number will be added to the end of the name to distinguish it.

Note 1: No actual editing takes place on the original compressed file...only the subsequent .wav file conversion.

Note 2: Drag and Drop is supported for this feature.

Large File Conversion to .wav (Expanded File Conversion)

Files in the Diamond Cut native format (.wav) are limited in size to 2.14 GBytes which is a Windows operating system constraint. At a 44.1 kHz sample rate with 2 channels (stereo) and 16 bit resolution, this

represents a time limit of roughly 3 Hours and 22 Minutes. If you are attempting to open a lossy file (such as an mp3*) that requires more than the 2.14 GByte .wav file size limit, your Diamond Cut software will deal with the situation automatically by creating multiple files. When this occurs, the first file created will be in the standard .wav format. Subsequent files will be of the Broadcast Wave Format (BWF) which is an extension of the .wav format. These expanded files are deposited in the same folder that held the original source file and they will have a sample rate of 44.1 kHz with a resolution of 16 bits. They will be named the same as the original file but having a _partXX added to the file name, where XX starts at 2 and continues up from that number. The Broadcast Wave format contains a field in its header which includes a starting time. When opening these files, they will have a start time that is correct with respect to the original lossy file; the subsequent Broadcast Wave files will not start at a time value of 00:00:00. If you want to edit time values that you have already created, you can view the time offset value in the Broadcast Wave header. To see that menu, it can be found at:

View->File Info

Next, choose the “Broadcast Wave Detail” button. You can change the header to adjust the offset time value of these files as needed. Note that this value is given (within the Broadcast Wave Dialog Box) in samples wherein the value is essentially the sample rate (44,100 samples per second) times the number of channels (1 for mono or 2 for stereo) times 2 bytes per word. If you change the header start time value, you must then press the “Update File” button for it to stick. To write the wave header information back to the file, you will have to close and then re-open the file to get the software to recognize the new time values.

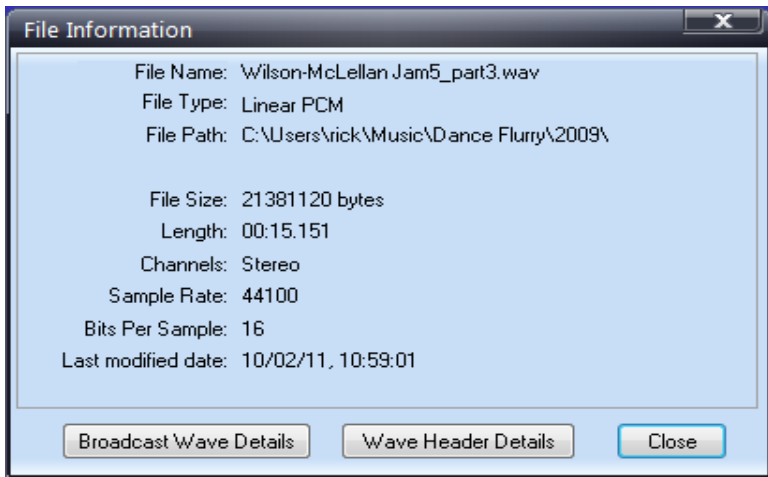


Figure 23 - The File Information Dialog Box

*Note: Only the mp3 lossy format is supported by the Diamond Cut Expanded File Conversion feature at this time.

Drag and Drop File Support

EIGHT/DC FORENSICS fully supports drag and drop file opening. You can drag a file onto the open program window or you can drag a sound file icon right on top of the EIGHT/DC FORENSICS Icon on your desktop.

Video/Audio Extraction System

View AVI Video and Edit Audio with AVI Audio Support

EIGHT/DC FORENSICS provides the ability to extract the audio portion of a video from the following file formats:

*.avi *.asf *.mpg *.mpeg

It converts the audio on these video formats into the .wav format. The Video/Audio Extraction System is found under the File Menu. To use it, just click on File/Open Source and in the Files of Type box select

Video Files and then navigate to your video file of interest. A video display box will appear as shown in the next figure:

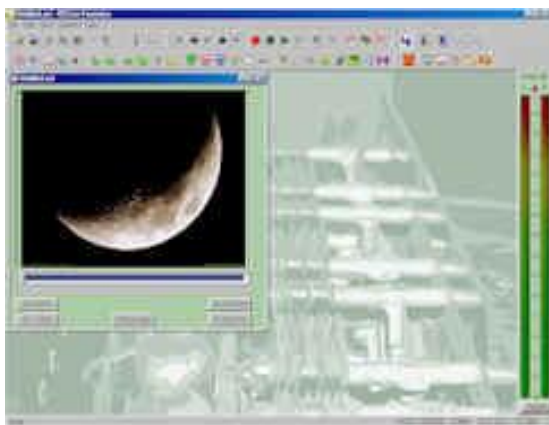


Figure 24 - The Video Extraction Window

To view the video and hear the audio, click the Play button on the Diamond Cut software toolbar (just like you would to play any audio file). You will see the video presented in the Video/Audio Extraction System display window and hear the audio via your soundcard and audio system. To stop or play the video, use the same buttons that you would use on audio files. The scrollbar at the bottom of the Video window indicates where you are in the process of the playing of the video relative to its total length. If you select the drop down "Time Display" on the View Menu, you can get more specific timing information regarding the Video file. There are several important controls below the scroll-bar, which are as follows:

- **Set Start Point** - Slide the progress bar pointer to the desired Start Point and click on this button. A marker will be dropped at the corresponding position.
- **Go To Start** - This moves the play pointer to the designated start position of the video.

- **Set End Point** – Slide the progress bar pointer to the desired End Point and click on this button. A marker will be dropped in the corresponding position.
- **Go to the End** – This moves the play pointer to the designated end position of the video.
- **Extract Audio** – This button is used to extract the audio from the video between your Start and End markers.
- **Progress Bar** – This is found on the lower status bar and indicates the progress of the Audio Extraction process.
- **Cancel Button** – This button is located directly to the right of the Progress Bar. Depressing it will cease the audio extraction process.

To operate the Video/Audio Extraction system, simply browse to the video file of interest and click on it. Set the desired start and end points with the play pointer. Click on the “Extract Audio” button. The progress bar will start to increase its value. When completed, the converted .wav file will be displayed in the Source Display window.

Note 1: The following keyboard keys perform some special functions when operating the Video/Audio Extraction system:

- *Home* – Go to the start of the file
- *End* – Go to the end of the file
- *Right arrow* – Go forward 1 second
- *Left arrow* – Go back 1 second
- *Shift + Right arrow* – Go forward 1 frame
- *Shift + Left arrow* – Go back 1 frame

Note 2: The Maximize button on the Video/Audio Extraction system is useful for increasing the size of the video frame so that it can be viewed in better detail. This is the square button in the upper right hand corner of the features display.

Save Source

This function saves the file to its current directory with its current name, etc. This menu item is only active in Fast Edit mode. In Classic Edit mode, files are automatically saved as you work on them.

Save Source As

“Save Source As” saves your Source file as it is, or assigns it a new name, sample rate, dithering technique, create .mp3s or .wmas. It can also be used to assign your file to a new location.

Your data can be saved in any of the following file formats:

- Wave Files (*.wav)
- BWF (Broadcast .wav)
- Vorbis (Ogg Vorbis) (Lossy Compression (.ogg, .oga)†
- WMA (compressed audio) Files
- MP3 Files (*.mp3) (provided you have a third party encoder installed)
- AIFF Files (*.aif)
- FLAC Files (*.flac) (Non-Lossy Compression)
- Compressed Formats (depending on what Codecs are available)
- Text CSV Files (*.txt)
- All Files (*.*)

Note: A dialog box will appear when you attempt to save your file in a compressed audio format such as .mp3 or .wma. From there, you can choose from a variety of compressed audio parameters.

†Note: For more information pertaining to Ogg Vorbis Lossy Compression support, please refer to the Ogg Vorbis Tutorial Section of this Users Manual.

Data Disc Burner



Overview

The Diamond Cut Data Disc Burner facilitates optical media storage using formats other than Red Book CD Audio†. You can archive data on optical media such as CD-Rs, CD-RW, DVD+/-R, and DVD+/-RW provided that your optical drive supports burning onto those formats. CD-Rs are capable of storing around 700 Mbytes of data while DVDs are capable of storing roughly 4,400 Mbytes (your mileage may vary). It works with most PC format data extensions including .wav, .mp3, wma, .jpg, and so forth. The Diamond Cut data disc burner has three points of user access. You can access it via the Data Disc icon on the toolbar, or you can access it from either the File or the CD Prep menu under the rubric of “Burn a Data Disc”. The Data Disc Burner supports complex hierarchical file tree structures having multiple directories and sub-directories. Disc layouts can be created either by browsing and adding or removing files directly from the Data Disc Burner dialog box or by dragging and dropping them into the Data Disc structure field directly from Windows Explorer. The file systems created by the Diamond Cut Disc Burner are ISO 9660 level 2 compliant. This system, however limits the number of characters assigned to a given file to 180 and the file extension limited to 3 with up to 8 directory levels. The Burner provides you with the option to choose the Joliet file system which provides for file names having up to 64 unicode characters (128 bytes) in length and directory hierarchies greater than 8 levels. Both Disc at Once (DAO) and Track at Once (TAO) modes of burning are supported, providing that your disc drive hardware supports it. Track at once mode provides the optional ability to keep sessions open after creating a CD session so that the disc can be added to at a later time.

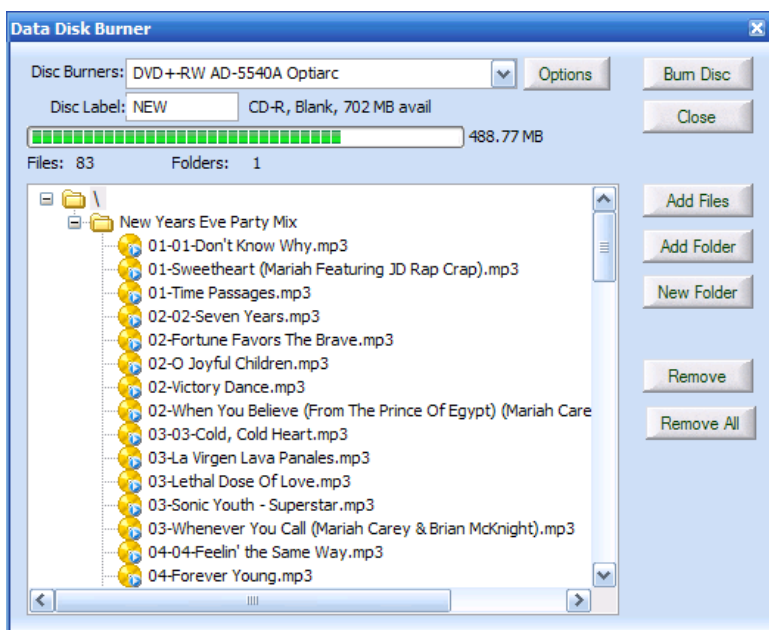


Figure 25 - The Data Disc Burner

The Data Disc Burner Controls

The following is a description of the various Data Disc Burner controls:

1. Add Files – Click on this button to add a file or multiple files to your disc layout from the Windows file system. First click on the location in the tree under which you desire the new files to appear. Then click on the Add Files button, browse to the file(s) of interest and then highlight them. Next, either double click on the file or click on “open”. The file or files will then appear in your disc layout field.
2. Add Folder – Click on this button to add an entire folder to your disc layout from the Windows file system. First click on the location in the tree under which you desire the new folder to appear. Then click on the Add Folder button, browse to the folder of interest and then

highlight it. To move the file into your layout, click on the “OK” button. The folder will then appear in your disc layout field.

3. New Folder – This allows you to insert a folder into your new file structure. To do so, first click on the location in the tree under which you desire this new folder to appear. Then, click on the New Folder button. A dialog box will appear and you can then “Enter a Name for the New Folder”. When you are satisfied with the name, click on “OK” and the new file will appear in the designated location in your disc layout. Files and other folders can then be added beneath that level in your disc layout.

4. Remove – This will allow you to remove a file or directory. Highlight the item in your layout that you wish to remove and then click on the Remove button. A dialog box will appear and query you if you are really sure that you want the item removed. Click on “Yes” or “No”.

5. Remove All – This allows you to remove the entire disc layout from your file tree except for the optical drive root directory. A dialog box will appear and query you if you are really sure that you want to remove the entire disc layout. Click on “Yes” or “No”.

6. Disc Burners Selector – Scroll to the Disc Burner that you want to use for your project.

7. Data Burners Options – Choose the setup that fits your needs:

A. File System Type

I. ISO 9660 (level 2)

II. Joliet File System (Long File Names)

B. Writing Mode

I. Track at Once (TAO)

a. Keep Session Open checkbox*

II. Disc At Once (DAO)**

C. Other Optional Attributes

I. Sort Disc Layout - When this is checked, the disc

layout is placed in alpha-numeric order.

II. Enable Optimal Power Calibration -When this is

checked, an extra step is added to the process to

optimize the burning power of the laser diode. This

creates a more consistent burn at the cost of extra

burn time.

*Note: Keep Session Open is only available in TAO Mode and for CD Burning.

**Note: Disk At Once is typically only supported for blank CD-R discs and rarely supported on DVD media. TAO mode is the preferred mode for creating DVD Video backups or data backups.

8. Drag and Drop – You can also drag and drop files, folders and directories directly from Windows Explorer into the Data Disc Burner.

Data Disc Burner Sample Procedure

Here is a fairly typical example of a procedure that you can use to burn a data disc:

1. Launch the Data Disc Burner.

2. Browse the “Disc Burners” feature to the Optical Drive of your choice.
3. Insert a media disc into that optical drive.
4. The top portion of the Data Disc Burner display should show the type of media inserted into the drive and also the number of Mbytes that are available for use.
5. Click on the “Options” button and check Joliet File System under the File System Type and then click on “Track at Once” under the writing mode option.
6. Use the Add Files, Add Folder(s) and New Folder features in order to create your disc layout in the workspace area found below the controls section of the Data Disc Burner.
7. When you have completed your layout, click on the Burn Disc button in the top right had corner of the application.
8. A progress bar will show you how things are progressing as the disc burns.
9. At the end of the burn process, the hourglass will remain showing while the system closes the disc. The progress bar will show a high percentage as this process proceeds.
10. The disc closing process could take as long as 15 minutes, so please be patient.
11. When the project has completed, the system will show a message which reads “Disc has been burned successfully. Please remove the disc from the drive.”
12. Done

†Note: Red Book Audio CDs can also be created with your Diamond Cut Software. This feature is accessed via the CD Button on the toolbar or via the CD menu item titled “Burn a CD”.

DC Tune Library



An Alternative Way to Focus the View of Your Audio Restoration Work

You have seen that the most typical way to view your work is via the **Classic-Edit** Source and Destination Workspace(s) as well as the **Fast-Edit** time domain display. Alternatively, the DC Tune Library can be made the primary focus of your Diamond Cut system by ticking off the “DC Tune Library” feature in the View Menu. Then, DC Tune library provides you with an alternative way to view all of your audio restoration projects and the subsequent archive that you will be creating. The DC Tune Library includes a full-featured audio file archive within the context of your DC8 software. It supports files in the .wav, .mp3, .wma, .ogg and .flac formats. It can search your hard drive and find all of your audio files and store their path, name, genre and other pertinent data within its file structure. After your DC Tune database has been constructed, it will provide you with lightening fast speed access to any of your audio files, whether they are in .wav, mp3, or .wma formats. You can search your database, sort it by various parameters and create and recall playlists. The DC Tune Library completely replaces the old Playlist feature in DC6 and prior versions.

You can play any file directly through the normal play features of DC8, or you can listen to a file or series of files by using the Preview feature found in the Diamond Cut filters and/or effects. You can even listen to the database by way of a complex series of filters and effects by previewing a file or playlist via the DC8 Multifilter. You can have the DC Tune Library start automatically by checking the “Start DC Tune Library” in the Preferences dialog (found under the Edit\Preferences Menu). You can also open this feature by going to the File menu, and then clicking on “Open DC Tune Library” and the spreadsheet view will appear. The system supports files having the .wav, broadcast wave, mp3, .ogg, .flac and .wma extensions. The system does not

support files having the .wma format that incorporate Digital Rights Management (DRM) schemes.

To create a library from the files that exist on a particular hard drive or sub-directory, simply click on “File” and then on “Add Folder to Library”. The preferences menu provides you with a number of options for obtaining information on a file, but .mp3 and .wma tag information will take precedence for those types of files. For files without tag information, such as Wave files, the Title, Artist, Album, and Genre information can be obtained from the directory structure of your hard drive. You can customize the way this information is used within the DC Tune Library’s by using the preferences dialog. By default it looks something like:

<Genre>|<Artist>|<Album>|<Title>

Use this system in conjunction with the highlighting and the “Move Up” feature to align the sorting to most closely resemble your database organization scheme. When adding a new folder containing multiple .wav files to the DCTune Library, individual track numbers for files contained therein will be set to the order in which they were originally created based on their time-stamps. If individual .wav files are added to the library (one at a time), they will appear in the DCTune database in the order in which they were added.

You have the option to customize the names applied to the various DC Tune Library columns by using the “Customize Column Header Titles” feature found under the Tune Library Preferences menu (or by clicking on a column title with the right mouse button and scrolling to the bottom item). Just click on the Edit button in the preferences and the following dialog box will appear:

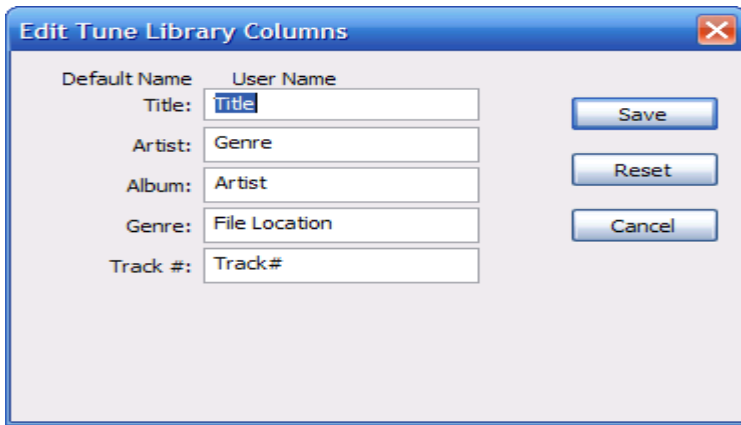


Figure 26 - Customize DC Tune Library Columns Dialog Box

After you edit the User Names, just click on Save, and the columns will become customized. Please note that this column customization feature can also be accessed via the right mouse clicking on any of the column headers.

The system will take some time (anywhere from 30 seconds to 20 minutes, depending on your system and the number of files found) to build your library, but once it is constructed, the system will no longer need to search your hard drive each time you want to access a particular file. If you have files on multiple drives, you will need to perform the creation step independently on each. Ultimately, all of the file(s) information will be integrated into one DC Tune Library. Note that the DC Tune Library does not copy or change any of your original files; it simply saves their location and other information into the DC Tunes database.

This database is stored in the following path(s):

- - - \My Documents\My Music\DC8\DcTuneDb

And a backup copy is saved as: -

- - - \My Documents\My Music\DC8\DcTuneDb.bak

The first path is the primary database, while the second one (.bak) is its backup. A Backup copy of the database is created each time you exit the program. The paths to the various files are stored in the DcTune Db

directory, but not the files themselves, which are all left in their original directories. Editing a file (like removing it from the DC Tune Library) will not delete the actual sound file from your hard drive.

After a period of time, you may want to update the database by re-doing the search process. Files that already exist will not be duplicated (if the appropriate “Tune Library” preference is selected); only the new ones will be added to the library.

If you have deleted files on your hard drive but not in the DC Tune Library you can update it by using the “Check Tracks” function under the “Tracks” menu. It will search for database entries that no longer have real audio files associated with them and highlight them for you.

If you ever decide to re-create a database from scratch, you can delete the DCTuneDb and the DcTuneDb.bak folders and the system will re-create these when you attempt to rebuild your DC Tune Library. When you build a DC Tune Database, you have the option to use file tags to place items into certain categories based on file tag information. You can enable the tags feature in the DCTune preference tab under the Edit Menu and are given several choices as follows:

- Use tags for track properties (MP3, WMA, flac, ogg)
- Use BWF Info (Broadcast Wave File)

The desired tag extraction mode should be established before building your DCTune database. If neither box is checked, no tag information will be used from any of your audio files.

The Library includes the following fields as viewed from left to right that can contain data concerning your audio files:

**Song, Artist, Album, Genre, Filename, Path, Track #, Length,
Type, Modification Date**

File	Artist	Album	Title	Genre	Year	Length	Modified Date
01 - Sound of Silence	Simon & Garfunkel	The Sound of Silence	The Sound of Silence	Rock	1966	3:57	10/10/2000
02 - Hotel California	Eagles	Hotel California	Hotel California	Rock	1976	6:30	10/10/2000
03 - Billie Jean	Michael Jackson	Thriller	Billie Jean	Pop	1982	5:55	10/10/2000
04 - Stayin' Alive	Bee Gees	Saturday Night Fever	Stayin' Alive	Disco	1977	5:13	10/10/2000
05 - Smiling Faces Sometimes	Simon & Garfunkel	The Sound of Silence	Smiling Faces Sometimes	Rock	1966	3:57	10/10/2000
06 - The Sound of Silence	Simon & Garfunkel	The Sound of Silence	The Sound of Silence	Rock	1966	3:57	10/10/2000
07 - The Sound of Silence	Simon & Garfunkel	The Sound of Silence	The Sound of Silence	Rock	1966	3:57	10/10/2000
08 - The Sound of Silence	Simon & Garfunkel	The Sound of Silence	The Sound of Silence	Rock	1966	3:57	10/10/2000
09 - The Sound of Silence	Simon & Garfunkel	The Sound of Silence	The Sound of Silence	Rock	1966	3:57	10/10/2000
10 - The Sound of Silence	Simon & Garfunkel	The Sound of Silence	The Sound of Silence	Rock	1966	3:57	10/10/2000

Figure 27 – The DC Tune Library Spreadsheet Display

A number of actions can be performed after your DC Tune Library has been built, including the following that are all available on the menu that appears when you click your right mouse button:

1. Display the Properties of a File
2. Play the Track(s)
3. Delete the Track(s)
4. Open the Folder for the track in question
5. Edit a Track using the Standard Diamond Cut features
6. Burn a CD from Selected Tracks
7. Create a Playlist from Selected Tracks

Additionally, you can search the database across all fields by using the “Search” field found in the upper left corner of the DC Tune Library. You can also perform a secondary sort by “Album” and then “Track” number when sorting on “Artist”. If a track number is not found in the sort, the system will resort to the use of the file modification dates to establish the order of the items in the display.

After your DC Tune Library has been built, individual file access will become virtually instantaneous simply by double clicking on the desired item. The highlighted file will play and the system will continue to play the entire database thereafter until you click on the

“Stop” button on the toolbar. The DCTune Library also includes a search engine. It will find any string of characters that you type into the search box by looking at the entire directory pathway for all matches. If a certain column is hidden, items contained therein will not be used in your search. You can see the relative play location of a particular file via the play progress bar (the horizontal green display). You can advance or retard the location of play by pointing and clicking to a different location on the play progress bar and the system will jump and then play from the new file location. As play progresses through the library, the playing item will become highlighted.

If you only want to add a single file to your DC Tune Library, click on “Add File to Library” and browse to the desired item. Further functionality can be found by right clicking on a file and then on properties which brings up the following screen:

The screenshot shows a 'Track Properties' dialog box with the following fields and values:

Field	Value
Title	Gene Krupa - Dark Eyes
Album	Krupa, Gene - Gene Krupa Jazz Trio (2 Songs)
Artist	1950s 45 RPM Records
Genre	Jazz
Track Number	1
Disc Number	345
Year	53
Length	3:29
File Type	mp3, 224kbps, 16bits
Preset to use for Multifilter playback	Very Scratchy Vinyl De-Clicker
File Info	Gene Krupa - Dark Eyes.mp3
Path	E:\Music\1950s 45 RPM Records\Krupa, Gene - Gene Krupa Jazz Trio (2 Songs)
Date Added	Thu Oct 01 12:17:22 2009

Buttons: Update, Close, Browse

Figure 28 – DCTune Library Right Mouse Track Properties Dialog Box

Playlists and Tunes from the library can be played via the Diamond Cut “Play” button, or via any of the Diamond Cut Filters and/or Effects “Preview” mode. The Multifilter “Preview” function is an especially

interesting way to listen to these files, since it allows you to stack up a series of filters and effects that you can apply to the playback in real time. To commence a preview after bringing up a particular playlist, make sure that the first line in the playlist is highlighted and then click on the “Preview” button associated with the filter or effect of interest.

The standard Windows mouse selection commands can be used on the DC Tune Database. If you want to delete a particular item from the database, right click on it and then right mouse to “remove”. If you want to remove a sequence of items, left click on the first item in the sequence, hold down the “Shift” key on your keyboard and then click on the last item in the sequence. Those items and all of the ones in-between will become highlighted. Then, right click on “remove” and they will be removed from your DC Tune Library. If you have a highlighted listing of files, and you want to un-highlight one item somewhere in the listing, depress the Ctrl key on the keyboard, and left click your mouse on the item of interest. To select all songs in the list, use the menu item, Edit/Select All.

Playlists are initially displayed in the order in which they are created. You can change the order by clicking on the desired column header of the DC Tune Library spreadsheet. Clicking on it once will put it in alphabetical order and clicking on it a second time will reverse that order, so you can have it ordered in either direction. Playlists can be re-ordered by dragging and dropping the songs from one position in the playlist to another. Note that only a Playlist can be re-ordered; if you have selected “All Songs” the order is determined by the sorted column.

Shuffle Play



Shuffle play is a form of random play of the selections presented to you in the DCTune Library Spreadsheet display area. Instead of playing the files in order, the system randomly advances through the listing. After a file (tune) has been played, it will be omitted from the selection process for the next round of random selection. Thus, no tune will be played more than once during a “Shuffle Play” session. This feature is located just to the right of the play progress bar and its icon looks something like the letter “X” but with arrows pointing towards its right

hand side. To activate this mode, click your mouse on the icon. When it is in its active state, it will present itself in two shades of green. When it is inactive, it will present itself in two shades of grey. Each time this feature is enabled on a listing (even if the listing is the same as a previous listing) the files will be played in a different (pseudo) random sequence.

Playlists

Playlists are used to further organize your tune library. A Playlist is a subset of the tune library based on your criteria such as Album or Artist or just picked at random. Playlists are not only used to play tunes but also to Burn CDs and Create Batches for the Batch File Editor. It allows a sequence of files to be played or transferred to an audio medium without having to manually cue up each audio file in real time. This is also useful and particularly advantageous in a forensics legal situation wherein an expert may be required to play certain groups of files in real time as an element of their testimony.

Multifilter Play Mode

Sometimes, it is desirable to apply filters or tone controls to the list of files that you are playing through the DCTune Library. The Multifilter checkbox in the DCTune Library facilitates this functionality. Every filter and/or effect in the Multifilter becomes accessible via this feature and is applied to the file being played in real time.

You can either use a selected preset, or use the currently selected preset when using the Multifilter for playback in the tune library. If you look at the track properties for any wave file, you can set the Multifilter preset that you want to use for that file. However, you also need to set the preference in the DCTunes preferences page to “use the preset from the tune library”. The default mode is to not change the preset. Please note that placing too many filters into the Multifilter in conjunction with the DCTune Library could lead to a situation in which the system “stutters” or “skips”. The reason for this in your CPU’s inability to keep up with the math required to process the data in real time. If this occurs, remove filters until the situation is resolved, or use a more powerful computer.

to the “Preset to Use for Multifilter Preset” listing and then scroll to the desired Multifilter preset (and settings) and it will be recalled in the future when that particular file is played. If a specific preset is not chosen, the DCTune Library uses the “TuneLibDefault” preset. To deactivate the Multifilter feature, click on the “X” symbol in the top right corner of the Multifilter. For more information, please refer to the Multifilter section of this Users Guide.

Creating a Playlist from your DC Tune Database

Use the general techniques described earlier to select a group of tracks that you wish to constitute a specific playlist. After you have completed the selection process, and while still pointing your mouse at one of the selections, right click and select “Create Playlist from Selected Tracks”. The system will choose the Album Name of the first item in the listing to use as the default the name for this playlist. If you do not want to use that name, you can simply edit it in the dialog box presented, as shown here:



Figure 30 – The Playlist Title Box

The playlist will be stored in .xml format in the following directory:

-- \My Documents\My Music\DC8\DCplaylists

The new playlist will now show up in the left hand margin of the DC Tune Database under “Playlists”. To play a desired playlist, click on the desired item, and then click on the “Play” button on the Diamond Cut toolbar. But, there are also a number of other options available to you on the menu which appears when you click the right mouse button on a particular playlist item:

1. Create New Playlist (using the existing one as the base)
2. Rename This Playlist

3. Delete This Playlist
4. Export This Playlist (dialog box provides export format alternatives)
5. Burn a CD from this Playlist
6. Run Batch Processing on this Playlist

You can also import pre-existing playlists having the following extensions:

.m3u,.wls (the older Diamond Cut Playlist Format), .cue and .pls. (These formats are sometimes referred to as M3U, WLS, CUE and PLS.)

To access this feature, go to the File Menu (of the spreadsheet display) and then to “Import Playlist”. The system will convert the imported format into the .xml format.

The system allows you to view and modify some of the properties of a particular file. To do so, right click on the file of interest in the DC Tune Library and then on “Properties”, and a dialog box will appear as shown below:



Figure 31 - The DC Tune Library Track Properties Dialog Box

You can modify information directly in the dialog box. When you are satisfied with your changes, click on the “Update” button. Additionally, you can use the “Browse” button to find a file that has been moved or renamed.

You can use this feature to update the DC Tune database and the MP3 or WMA tag information for the selected files. If multiple files are

selected, some fields are disabled such as the Title and Track Number. You can enter information for the group of files such as the Album Title and update all of the selected files at once.

CD Creation / CD ROM Creation / CD ROM Burning / CD Burn

In much the same way that you can construct a playlist, you can also set up a listing to create a Red Book CD. This feature is accessed via the right mouse button or the CD icon, and details can be found pertaining to its operation in the **CD Burner** section of this User's Manual found under the **CD Prep** section. It is worth noting that you can drag and drop items from the DC Tune Database directly into the CD Burner Dialog box.

Playlist Export Feature

Your Playlist can be exported into other programs that permit importing data such as disc label makers and CD-ROM writing programs. The feature will export the following parameters:

1. Title Number in sequence
2. Title of Selection
3. Length of Time of Selection
4. Total Length of Time of the Selections

Clicking on the Export button found on the right mouse button activates this feature which allows you to save it in a number of formats. When you click the Export List button, you can specify a file name. The default will be saved in the *.txt format which can be opened by many other programs. It can also be saved in your choice of other formats including .m3u, and .cue and .pls(sometimes referred to as M3U,CUE, PLS).

.wma File Encoding

You can encode a file in compressed .wma format if you so choose. To do so, click on .wma and you will be presented with the following dialog box with a list of compression options from which to choose. In general, higher bit rates (kbps) produce better sounding files at the expense of greater memory space requirements. You decide which options best meet your needs.

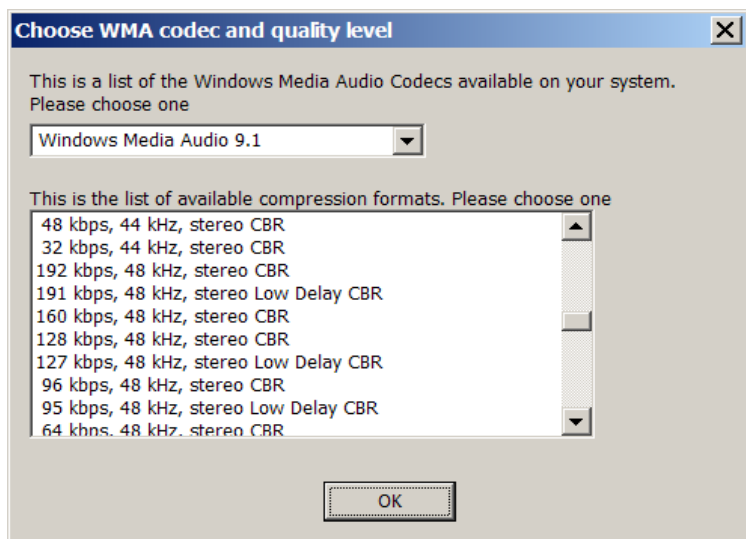


Figure 32 – The WMA Dialog Box with Compression Options

To view the other compressed formats that are available, click on Compressed Formats in the “Save as type” field and highlight “Compressed Formats”. Then, click on the “Available Audio Codecs: Please Select One” field. A listing of available formats will be indicated. The contents of this listing will be dependent on your operating system and/or which Codecs you have installed.

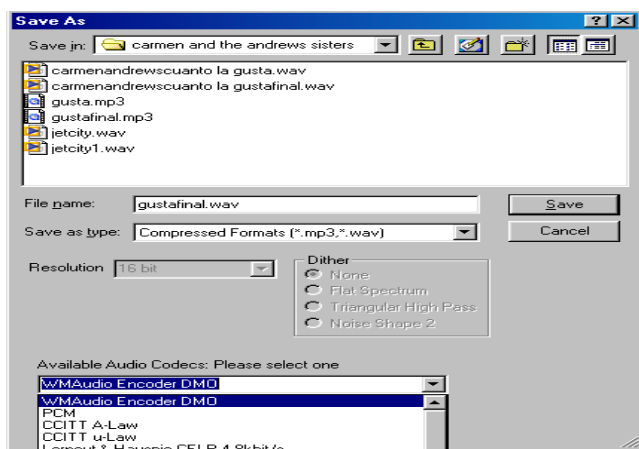


Figure 33 - Save As Dialog Box

EIGHT/DC FORENSICS has the ability to use an external Encoder to help you turn any .wav file into an MP3. This process simply requires that you download one of these popular encoders, go to your Edit/Preferences/Mp3 Encoder and define a path to that particular .EXE file. From that point on, if you choose to SAVE AS or Batch File Processing and want the end result to be MP3 files, DC8/DC FORENSICS will use that encoder to make the conversion. We recommend the LAME encoder as the most popular and highly regarded.

Playing CDs using the DCTune Library

You can play CDs (Red Book Audio) using the DCTune Library. Simply place your CD into your optical drive. After the drive “spins-up” (which will take a few seconds) double click on the words “Audio CD” which is in the “tree” listing within the left-hand column of the DCTune Library display. You should then see the following appear as part of the tree:

“Audio CD: xx:yy”

where xx:yy is the length of the CD in minutes and seconds. Single clicking on “Audio CD: xx:yy” will then construct a listing of the tracks which will appear in the right-hand display of the DCTune

Library. To play these CD tracks, just click on the play button located on the tool bar. Most of the various controls previously described and associated with the DCTune Library also apply to the CD Player (including Shuffle Play). However, the Multifilter does not work in conjunction with the CD player.

Import Playlists

This feature allows you to search your hard drive and import playlists into your DC8 software. It will search your hard drive for playlists in the .wls, .m3u and .cue formats and convert them into the .xml format for compatibility with the DC8 system.

Close Source

This command closes a previously opened Source and/or Destination .wav file.

Rip CD Tracks

Rip CD Tracks to .wav, or .mp3 or .ogg

This feature allows you to “Rip” (convert) CD Red Book Audio CDs to Wave files (.wav) or MP3s (.mp3) or Ogg (.ogg). To operate this feature, first place the CD into the CD ROM drive in your computer and then launch the “Rip CD Tracks” feature found under the File Menu. A dialog box will appear with the following selections and features:

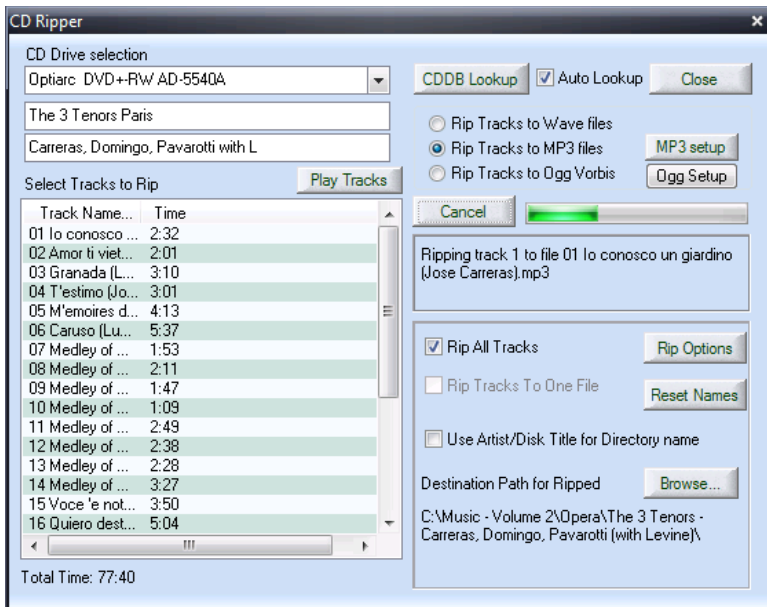


Figure 34 – The CD Ripper Dialog Box

- **CD Drive Selection:** Used to choose the CD ROM Drive that you want to use to rip your CD.
- **Artist Field:** Allows you to identify the Artist for the Ripped File
- **Title Field:** Allows you to identify a Title for your Ripped CD.
- **“Rip All Tracks”** precludes the necessity to highlight the desired tracks to be ripped from the CD.
- **Select Tracks to Rip Box:** This box shows all tracks which are available on the CD either as Track Numbers or Track Names depending on whether or not the CDDb had been utilized. Selecting Tracks to be ripped is accomplished as follows:
 - To select a single Track to be ripped, simply point your mouse towards the desired track, and use the left mouse button.

- To select Tracks at random, point your mouse towards the track(s) of interest and use the Ctrl Key in conjunction with the left mouse button.
 - To select a range of Tracks to be ripped, first point your mouse at the bottom most track of interest in the list and use the Ctrl Left mouse button to highlight the same. Next, point the mouse to the top Track in the range of interest and click on Ctrl + the Left mouse button. The range of tracks between these two Tracks all will become highlighted.
 - To clear all selections, point the mouse to the right hand side of the Track selection box and left mouse click. All Tracks will become un-highlighted.
- A checkbox is provided for CD jitter correction. Potentially this can improve the accuracy of the transfer process, with the tradeoff of a slower ripping speed.
- CD Database Automatically assigns track names to inserted CDs
- CDDDB Lookup: CD Data Base Lookup correlates the tracks on your CD to Song Titles that can be found on the Internet. To use this function, first you must place the CD in the CD ROM Drive. Next, launch the “Rip CD Tracks” feature. After the track list is developed and you are signed onto the Internet, clicking on the CDDDB Lookup Button will find (if available) the CD title and song list and convert the tracks to song titles. This process will take place automatically by selecting the Auto Lookup checkbox, but will only work if you are connected to the internet at the time of the CD rip. If you want to edit/change the name of a particular track, use the Windows slow left-mouse double-click on the track of interest (double click with about 1 second between clicks). It will then become highlighted, allowing you to change the track title. Setup for the CDDDB system can be found in the Preferences section of the Edit menu under the “CDDDB Setup” tab. For more details regarding CDDDB setup, please refer to the preferences section of this manual.
- Selector: Choose between the following:
 - Rip Tracks to .wav files

- Rip Tracks to MP3 Files (You will need to install an external encoder for this feature to work. Please refer to the MP3 Preferences Setup section of this manual)
- Rip Tracks Button: Click on this button to start the “Ripping” process.
- Ripping Status Box: This is located directly below the “Rip Tracks” button and includes the following:
 - The system will indicate if the System is “Ready.”
 - The system will indicate Error Messages
 - The system will indicate the progress of the ripping process via a “Progress Bar.”
 - It will indicate when the System is “Done.”
- Modes of Operation Checkboxes:
 - Use Artist/Disc Title for Directory Name
 - “Rip All Tracks to One File”: This can be useful in situations wherein no dead-time is desired between contiguous tracks or when you want to process all the files on a CD through a filter in one single operation.

Destination Path for Ripped Files: Browse to the desired Directory

An MP3 Setup button is provided to quickly bring up the MP3 encoder(s) preferences box should you desire to make some changes in its settings, precluding the necessity of having to go to the Edit/Preferences menu item.

Note 1: ID3V2 mp3 tags are automatically added by the ripper to MP3 files. These include the Title, Artist, Album, Track #, Maximum tracks and Genre.

Note 2: Ripping directly to an external (USB or Firewire) drive is not recommended. Timing errors in the data pathway can result in the creation of corrupted files. If you want to maintain ripped files on an external drive, rip them to a directory on your C drive first and then transfer these files to your external drive(s) thereafter.

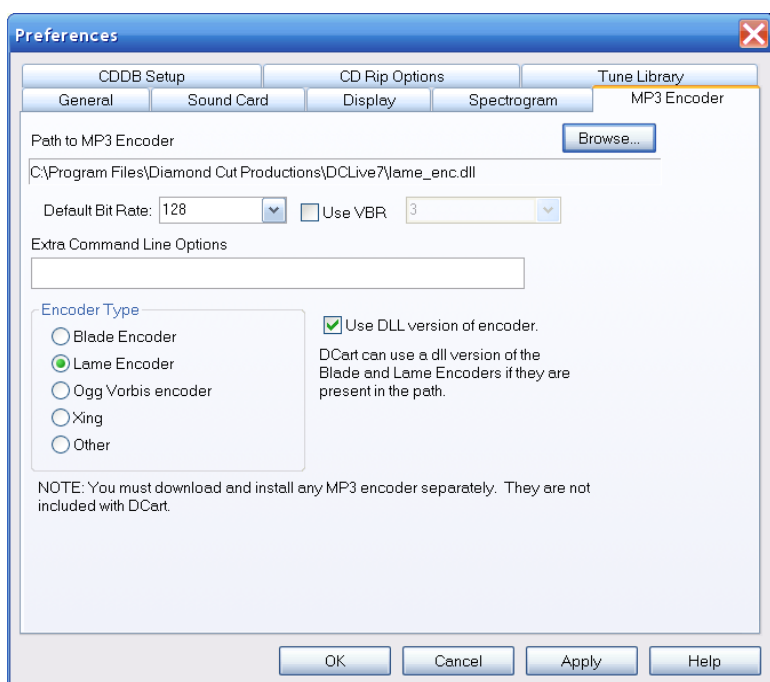


Figure 35 – MP3Encoder Setup

CD Ripper Preferences

If you click on the “Rip Options” button, The CD Ripper Preferences Menu will appear .

File Options

This controls the file overwriting and opening files options.

The file format extension that is created by the “Create Playlist after Ripping” feature is .XML.

CD Control Options

One feature involves limiting the ripping speed of the system. This is useful sometimes when a CD ROM has become warped due to its paper label or other variables. Disc “warpage” can create a situation in which the CD ROM will not rip, but slowing the disc down can sometimes overcome this difficulty. Also, it is generally good practice to leave the

“Jitter Correction” on, but it substantially slows down the ripping process due to the extensive calculation required to perform the task. But sometimes better CD ROM ripping success may be had with this feature turned off when encountering extremely problematic discs.

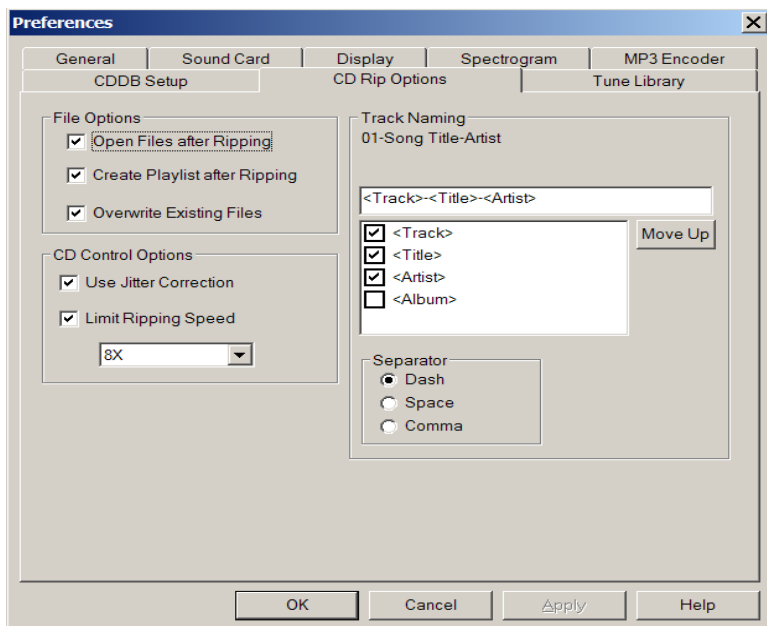


Figure 36 – The CD Ripper Preferences Menu

Track Naming

You control how the resulting ripped tracks are named via the Track Naming feature. Typically track names are simply the track # and song title, but many folks prefer to include more information in the title.

It derives its information from the CDDB or via manual data entry in the Ripper and constructs the file name accordingly. Select the fields you want included in the name and arrange them by using the “Move Up” button. Four choices are provided and 3 possible data separators including a Dash, Space or a Comma. To change the order, click on one of the attributes and then click on “Move Up” button. That will

cause that item to be moved one position to the left in the hierarchy. Repeat the process until you obtain your desired final result.

Open Destination

This command allows you to define the name and desired storage location of the processed version of the .wav file that you are about to create through the use of the various signal-processing tools of the DC8/DC FORENSICS. The use of this command is optional since the software creates temporary files automatically.

Save Destination As



Since DC8/DC FORENSICS does not require the Destination workspace file name to be defined before your audio processing session, this command is used to define a Filename and directory location for your Destination file following the completion of an audio processing session, should you desire to save it. This function can also be activated by using the file folder icon in the File Tool bar of the workspace or by using the **Ctrl + S** hot key. Note: It is almost never necessary to manually save a file in Classic Mode. The program automatically does this for you.

Close Destination

This command allows you to close a file that has just been processed from the Source file. The working Destination file will have been stored on a temporary basis in a temp.wav file. When you attempt to close the Destination file, you will be prompted to indicate whether or not you want to save it. If you do, then you will be prompted to define a path and a name for your processed file to be saved in.

Clone Source

Quickly copy a.wav file from the Source to the Destination window

This feature takes whatever .wav file exists in the Source Window and replicates it in the Destination Window. It is only applicable in the

Classic Edit mode and not the Fast Edit mode. It is particularly useful when operating the system in Selective Filtering Mode.

Make Destination the Source

This command takes the file that has just been processed, and makes it the Source file in a new workspace window. This is a useful feature, since most sound jobs require several passes utilizing several different signal-processing techniques to affect a complete audio restoration. When using this command, the program will prompt you to name the file. It is recommended that you accept the suggested name. The original Source file may be deleted when making the Destination the Source by using the appropriate checkbox in the dialog box. *This command is grayed out until you have a Destination file.*

Delete Files

This feature allows DC8/DC FORENSICS to delete a file from a hard drive. Since .wav files tend to be large, this command will be used often. The software will prompt you to be sure that you want to delete the selected file before doing so. Remember that every minute of stereo audio sampled at 44.1 kHz consumes 10.584 Mbytes of disc space, which is useful to know when it comes time to clear up some disc space in order to get ready for your next sound restoration job.

Deleting a Wave file (Tutorial)

1. **Warning! This operation cannot be undone!**
2. Click on "File" and a pop down window will appear.
3. Click on "Delete File" and the Delete File Dialog Box will appear.
4. Choose the Drive and the Directory from which you desire to delete a .wav file.
5. Click on the Filename that you desire to delete in the Filename field. *
6. Another dialog box will appear, inquiring whether you are sure that you want to delete the chosen file.
7. If you click on "yes", the file will be deleted.

8. If you change your mind, and click on "no", the DC8/DC FORENSICS program will revert back to its initial window, and the file will not be deleted.

* **Note 1:** Multiple files that are sequential in the file listing can be deleted in one operation by clicking on the first item on the list, and then dragging the mouse pointer down to the last file you desire to delete. The files that are about to be deleted will be highlighted.

***Note 2:** Multiple files that are not sequential in the file listing can also be deleted in one operation by holding down the CTRL key at the same time that you click on the appropriate item you wish to delete with the left mouse button. The files that are about to be deleted will be highlighted.

Print



This commands prints the screen as defined by your choices identified in "Print Setup." It prints the present DC8/DC FORENSICS screen as "WYSIWYG" (What You See Is What You Get). Note that the screen is a very complex graphic with high-resolution images for Forensics applications. As a consequence, some laser printers will exhibit a problem using the Print command because of their rasterization demands – this just means your printer does not have enough internal memory and would be unable to print any graphic of this complexity. This is not generally the case with inkjet printers. This function can also be activated by using the Printer icon in the File Tool bar of the workspace or by using the **Ctrl + P** hot key.

Printing Help-File Topics

Sometimes it is useful to be able to read Help file topics from paper rather than from your computer screen. This is accomplished in the following manner:

1. From the Help file, select the topic you are interested in printing.
2. Click on the "File Menu" with the left mouse button.
3. Click on "Print Topic".

Printing a Screenshot (Tutorial)

Though DC8/DC FORENSICS now includes a Print function, you may still find the need to print a specific multi-layered screen.

1. When you've focused in on the screen sector that you want to print (i.e., a filter or dialog box), simply hit Alt and Print Screen simultaneously. This will copy the image into your Windows clipboard.
2. Enter any paint-type program or even a word processor that accepts graphics and hit Ctrl V or use the Paste tool from the Editing menu.

Note: If you need to print the entire screen, just hit the "Print Screen" function on your keyboard. Do not use the "Alt" function.

Print Preview

Simply shows a snapshot preview of what you're about to print.

Print Setup

This command opens the Print Setup dialog box in which you can define the following parameters:

1. Choose the Default Printer or choose some other printer
2. Choose the orientation of your printout sheet:
 1. Portrait (This orients the paper vertically)
 2. Landscape (This orients the paper horizontally)
3. Choose the paper parameters that you desire:
 1. Size (Default value is 8 1/2 inches x 11 inches)
 2. Paper Source (Choose between the paper cassette or manual feed)
4. Properties: This allows you to select amongst the various printer dependent parameters provided by your specific device.

Page Setup

This feature allows you to select among the following printing choices:

1. Include File Information in printout (File name and Date of last modification)
2. Print the Source Waveform
3. Print the Destination Waveform or Spectrograph
4. Create a Page Title up to 80 characters long (the default title is “Diamond Cut Audio Restoration Tools”). To enter your title, just type over the default title. The maximum length that will appear on your printout will depend on the font that you choose, smaller fonts allowing longer character strings. To define the font size, go to the Preferences menu. Under Preferences, choose the “Display” tab. Under the “Display” tab, you will find a field in which you can enter an integer value for “Display Font Size”.
5. Set Margins Spacing

Exit

Note: Exiting the program will also clean up all temp files in Classic Mode. It will ask you if you want to save any unsaved files before you exit.

The Edit Menu

Undo

This feature allows you to return to a previous version of a destructively edited file after using such features as Mute, Fade-In, Fade Out, Cut, and Copy / Paste / Insert. After an “Undo” is performed, it is removed from the “Undo” listing. You can also access the function using the **Ctrl + U** Hot Key.

All single file operations can be undone by using the Undo menu item. The default number of undo levels is 10. The number of undo levels is selectable in the Preferences dialog box. When you close DC8/DC FORENSICS (Exit the program) all undo information will be lost. Many Edit Menu functions will not appear in the list if no file has been opened.

Undo Procedure Using Classic editing mode (Tutorial)

1. Click on the "Edit Menu."
2. Click on "Undo."
3. Click on the menu level of undo that you desire. The top "undo" in the stack represents the state of the .wav file prior to the latest operation which you performed, and the next down in the stack is the one before that, and so on. Variable levels of undo are provided, and are settable under the Preferences section of the "Edit Menu."

Undo Procedure Using Fast-Edit Mode (Tutorial)

Use the Fast-Edit history window to temporarily go back to the last step. This method does not permanently undo the last action. Use the right mouse button to delete the last operation. Double click or use the right mouse button menu in the Fast-Edit history window to delete the last action or series of actions. You will always be prompted by the computer to make sure this is what you want to do. This operation does not display the undo levels in the Edit/Undo Window as the Classic mode does.

Copy



This is the key with an icon consisting of two paper documents contained within its perimeter. It is used to Copy a highlighted selection of a .wav file from either the Source or Destination .wav file, and place it onto the program's clipboard. This function can also be accessed by using the Copy Icon in the upper left tool bar or by using the **Ctrl + C** Hotkey. You can also access Copy and several other functions by selecting an area and using the right mouse button menu.

Copy and Paste Procedure (Tutorial)

1. Determine the section of the .wav file that you wish to copy for the "copy and paste" operation. *You may use the Zoom-In feature if you would like to copy a very small portion of the Wave file.*

2. At the beginning of the section of the .wav file which you desire to copy, click down on the left mouse button and keep holding it down as you drag the timing bar (with the mouse) towards the right of the workspace.
3. Stop dragging the mouse at the end of the desired "copy" sector of the .wav file.
4. Release the left mouse button. You will notice that the sector between the two timing bars will remain highlighted in yellow. This will be the "copy" sector.
5. Click the right mouse button anywhere in the workspace area and a pop-up window will appear, providing you with three choices.
6. Click the right mouse button on "copy" and the highlighted sector will be transferred to a temporary storage location on your hard drive.
7. After the transfer is complete, highlight the area in your .wav file where you desire to "paste over" the previously copied segment using the mouse drag procedure outlined in steps # 2 through 4.
8. Click the right mouse button again anywhere in the workspace area.
9. This time, click on "Paste Over."

Important Note:

The timing rules used by "Copy and Paste Over" are as follows:

1. Copy and Paste Over operations always begin at the leftmost timing marker of the highlighted area of the workspace.
2. If the "Copy" sector is shorter than the "Paste Over" sector, the length of insertion is determined by the length of the "Copy" sector of the .wav file.
3. If the Copy sector is longer than the "Paste Over" sector, the length of insertion is determined by the length of the "Paste Over" sector of the .wav file.

Manual De-Click with "Copy" and "Paste Over" (Tutorial)

1. Listen to your .wav file and determine the location of the click, pop, or thud that you desire to eliminate.
2. Zoom-In on the section of the .wav file containing the click using the feature having the same name.

3. Continue Zooming-In alternately listening to the .wav file until you see the troublesome artifact in the DC8/DC FORENSICS workspace. It will take some training to be able to identify transients visually, so be patient during your learning curve.
4. Using the left mouse button, highlight a sector of the .wav file just prior or just after the transient event, being careful not to overlap the highlighted sector onto the actual transient. The highlighted sector must be at least as long (or longer) as the transient event.
5. Click on "Edit."
6. Click on "Copy."
7. Using the left mouse button, highlight the transient event itself.
8. Click on "Edit."
9. Click on "Paste Over."
10. Zoom back out and listen to the .wav file.

Important Note:

The replacement algorithm used in the Impulse Noise Filter is much more sophisticated compared to one used in this manual de-clicking procedure. Whenever possible, you should use the Impulse Noise Filter to de-click a record. Only in the unusual or extreme case wherein the Impulse Noise Filter has been unable to automatically remove a particular artifact, should you use this manual process.

Cut



Just as the title implies... Cut allows you to highlight an area and then cut it out of the file. This feature is useful when it is necessary to reduce the musical portion of a segment for a competitive event in which the total length of the program is governed, and you do not want to eliminate either the beginning or the end of the song to achieve that end. You can also access this function by using the **Ctrl + X** hotkey.

Splicing out a portion of a Wave file (Tutorial)

1. Highlight and play the portion of the .wav file you believe that you would like to splice (cut) out.

2. Play the sector (using the Play button on the toolbar) to make sure that you have identified the correct timing for the segment you wish to remove. Re-highlight the correct area if necessary.
4. Click on the Edit Menu.
5. Click on "Cut."

Important Note:

Fast Edit Mode can perform multiple cuts on even very long files almost instantly.

Manual De-Clicking with Cut (Tutorial)

1. Zoom-In on the area of interest in the .wav file.
2. Highlight the click or pop impulse using the mouse. This should be a very small segment of the file.
3. Click on the Edit Menu.
4. Click on "Cut".

Important Note:

Removing clicks by cutting, although very easy to use, is not recommended because it actually shortens the total program length from the original. Instead, consider using the manual interpolator "I" key.

Paste

As the name implies and similar to every word processor you've ever used, Paste allows you to take whatever element you've cut or copied and put it back into your current file in one form or another. We've created many different ways to paste your material. We've thought of everything but Paste Eat, which we've reserved for the Elementary School edition of the product.

Append to End

Paste Clipboard Contents Directly To End of a File

The "Append to End" feature takes whatever file or portion thereof you have copied onto the clipboard and attaches it to the end of the

displayed .wav file. The resulting file will become larger so you may have to zoom-out to see it in its entirety.

Insert at Start

Paste Clipboard Contents Directly To Beginning of a File

The "Insert at Start" feature is the complimentary function to the "Append to End" feature. Simply stated, it takes whichever file has been copied to your clipboard and attaches itself to the beginning of the displayed .wav file. As a result, the total newly formed file length will become larger so you will have to zoom-out to see it in its entirety.*

Paste Interpolate (Time Domain Technique)

"Paste Interpolate / Time Domain" allows you to manually correct a recording impulse noise defect such as a tick, pop, click or thud. Some folks fondly refer to this technique as the "Oterpolator" named after the "O" hot-key that activates it. This uses time domain interpolation techniques to calculate the replacement signal and is primarily optimized for short time interval situations.

To use either of the Diamond Cut two Paste Interpolation functions, simply highlight the area in the Source file in which you are observing a noise event. Next, click on the Edit Menu, and scroll down to "Paste". Lastly, click on the Interpolate feature of your choice, and the event will be replaced with a new waveform. This new waveform is calculated by a high-order modeling algorithm utilizing up to a maximum of 10,240 samples of data (up to around 0.232 seconds on a 44.1 kHz sampled file {232 mSec}). You can also access this manual interpolation function for both channels by using the "O" Hotkey.

Manual De-Clicking with Paste Interpolate (Tutorial)

1. From the Source Workspace, listen to your .wav file using the Play feature, and determine the approximate location of the click, pop, or thud that you desire to eliminate.
2. Zoom-In on the section of the .wav file containing the click using the feature having the same name.
3. Continue Zooming-In, alternately listening to the .wav file until you see the troublesome artifact in the DC8/DC

FORENSICS workspace. It will take some training to be able to identify transients visually, so be patient during your learning curve.

4. Highlight the transient event with the mouse drag procedure.
5. Depress the "O" key on your keyboard. The transient will be replaced with new signal approximating the audio waveform that should have been there.

Note 1: If you are interested in interpolating only the Left or Right channel with the Time Domain routine (rather than both channels together), just use the L or R channel selector buttons on the toolbar. You must be using WDM drivers, however, for this method to work; it will not work with MME drivers.

Note 2: Other methods of manual interpolation include Paste Interpolate (Bi-Modal Technique) and Direct Spectral Editing. Please refer to those sections of this user's guide for details.

Paste Interpolate (Bi-Modal Technique)

DC8 provides you with an automatic switchover method for inserting manual interpolations onto a waveform called "Paste Interpolate". It is used in the same manner as the standard Time Domain Paste Interpolator. This bi-modal interpolation technique uses a combination of time and frequency domain algorithms to determine the best replacement signal and is better optimized for longer time interval interpolations. It automatically converts to a frequency domain system when the selected area of the waveform is greater than a few milliseconds in length. Along with the capability of fixing very large impulse noise events (events which are very long and tall) with improved accuracy, it can also be used to remove other recording anomalies such as coughing or chair movement from a recorded concert performance. Additionally, it can be used to correct things like "fret" noise from a studio recording of a guitar and other similar studio related problems.

Its interpolation capability has been lengthened and its accuracy improved on long events compared to the older Diamond Cut time domain only counterpart. Previous versions of the Paste Interpolate function utilized only time domain curve fitting techniques. This version now uses time domain as well as frequency domain

interpolation techniques. This combination of techniques results in more accurate interpolations, especially on longer time interval events. From a user's perspective, it works in the same manner as the time domain Paste Interpolator, and uses the original Diamond Cut "I" Hotkey. If you want to interpolate the Left Channel only, use either the "J" or the "SHIFT + I" Hotkeys. Conversely, if you want to interpolate the Right Channel only, then use either the "K" or the "CONTROL + I" Hotkeys. It is capable of interpolating up to 0.5 seconds of a .wav file. It automatically switches from a time domain interpolator to a frequency domain interpolator on highlighted signals which exceed several milliseconds.

Note 1: Generally speaking, the best interpolations for short impulsive events (250 samples and below) are achieved via the Time Domain Interpolator routine (the "O" key)

Note 2: Generally speaking, the best interpolations for long impulsive events (250 samples and above) are achieved via the Frequency Domain Interpolator (the "I" key).

Note 3: Notes 1 and 2 are not hard and fast rules and the best interpolation results are due to a combination of the method and the nature of the audio material that you are working with. When in doubt, try the "Direct Spectral Editor" (found under the Edit Menu) which uses only frequency domain techniques. Compare its result with the result produced by the "I" key and its affiliates.

Note 4: Other methods of manual interpolation include Paste Interpolate (Time Domain Technique) and Direct Spectral Editing. Please refer to those sections of this users guide for details.

Paste Over



Paste Over allows you to insert the portion of the .wav file located in the clipboard over the top of a different location in your .wav file or to other .wav files. (This operation will delete the portion of the .wav file that previously had been in the particular location, installing the temporary file in that position instead.) The "Copy and Paste Over" feature in DC8/DC FORENSICS can also be used to manually "de-click" or "de-pop" a sound source (see tutorials under "Copy"). This function can also be accessed using the **Ctrl + V** Hotkey or by right clicking the mouse after you've copied or cut a portion of the .wav file.

For various tutorials of Copy and Paste Over, you can refer to the Copy section of this manual.

Paste Insert

Unlike Paste Over, Paste Insert does not wipe out the sector of the .wav file where you desire to place the contents of the “Copy” temporary file. Instead, it inserts the area you are pasting into the targeted area and just moves the existing material over.

Paste Mix

Paste Mix allows you to add or “mix” one file (or a portion thereof) to a second file. This feature is useful for creating “voice-overs,” or inserting special effects on top of a previously created sound track. This feature works in conjunction with the Copy function. In many cases it will require that two files be opened, one in the Source Workspace, and a second in the Destination Workspace. But this is not mandatory in that you can “paste mix” a portion of a file back onto itself if desired. The file that you open in the Source Workspace can be the file onto which you will be mixing. The File which you will be establishing as the “voice over” or special-effect, might be the one opened in your Destination Workspace. In other words, you can mix the Destination file into the Source File, in this example. The process can also be performed in reverse, wherein you can mix a portion of the Source file into the Destination File. These processes are undo-able, so that you can experiment until you are satisfied with the result. To use this feature, you will be highlighting the portion of the Destination File that you want to mix into the Source file. You will then use the Copy command to place it on a clipboard. Then you will highlight the Source file location in which you want the voice-over mixed in. When you run Paste Mix, you will be able to adjust the Source and Destination gain settings over a range of from +12 dB to – 100 dB.

Paste Crossfade

Paste Crossfade is the cousin of the Paste Mix feature. It operates in a similar manner, with the difference that there is a time varying function applied to the gain settings, so that a “cross-fade” effect can be produced. This feature is useful when you want to fade one song (or

file) into another, with no “dead-air” in between. When you run Paste Crossfade, you will be able to adjust the File-1 (clipboard) Start and Stop Gain settings as well as those for the target file (File-2). You have available four gain controls in total. You will also be able to control the dynamics of the cross-fade by selecting Linear In, Log In, or Log Out. This feature is undo-able when executed under the Edit Menu. The “Crossfade” feature is also available under the Filter Menu.

Using the Paste Cross-fader (Tutorial)

This method of cross fading is undo-able.

1. Open the File (song) into the Source Workspace that you desire to be the first in your cross-fade timing sequence.
2. Open the File (song) into the Destination Workspace that will be the second song in your cross-fade timing sequence.
3. Highlight, using the mouse drag procedure, the file to which you will be cross-fading in the Destination Workspace. You must highlight the file from the point of segue all the way to the end of the song, assuming that you desire to maintain the entire song in the sequence.
4. Click on "Copy" under the Edit Menu. This procedure may take a bit of time as this relatively large file is copied onto the clipboard.
5. Next, highlight the end of the file located in the Source Workspace. The area that you highlight will determine the cross-fade timing interval. The longer you make this interval, the slower will be the cross-fade sequence.
6. Click on "Paste" under the Edit Menu.
7. Next, click on "Paste Cross-fade" under the paste menu.
8. Set the gain controls in the dialog box as follows: (default values will work)
 - File 1 Levels:
 - Start = -100 dB
 - Stop = 0.00 dB
 - File 2 Levels:
 - Start = 0.00 dB
 - Stop = -100 dB
9. Choose the cross-fade timing that you desire. Linear usually produces a pleasing effect.
10. Click on "OK"

11. After the processing has been completed, you may click on the play button on the Toolbar to hear your results.

Paste As A New File

This feature provides a convenient means to chop a large file into smaller pieces, and assign new .wav file names to these subset files. It is useful for creating a number of .wav files that could be listed and quantized for CD-R indexing from a single large file such as that which you might have from transferring a Vinyl LP, or a concert tape recording to DC8/DC FORENSICS. A popup window will appear in which you can redefine a file name for each “chopped” file subset.

Paste Silence

Simply insert a predetermined amount of silence in your .wav file and select from the Beginning of the file, End of the File or the Beginning of a selected area.

Paste Bleep (tone)

Insert A Tone To Cover Objectionable Material

The Paste bleep tone feature interjects a 440 Hz Sine wave (A above middle C) at a -10 dB level over the highlighted area of a .wav file. It is intended to be used to “bleep out” unwanted verbiage or other sounds. To use this feature, merely highlight the area of the .wav file in the Source Window that you want to “bleep out.” Then, navigate to the Edit/Paste/Bleep (tone) feature. The highlighted area of the .wav file will be replaced with the bleep tone. This is an undoable function.

Select All

This command selects the entire .wav file. It can also be activated by using the **Ctrl + A** Hotkey or a double left click within the file area. If you have markers present in the file, double clicking will highlight the area between the two closest markers.

Pencil Editing



This device is provided for single sample editing of your waveform. Though not as useful for general waveform drawing, it can be very useful for eliminating tiny clicks and pops that will show up if you are zoomed in closely on the waveform. It will not be active until you've zoomed into this level. You can access this feature by using the Pencil icon located in the toolbar or using the **Ctrl + E** Hotkey.

Remember, you must be zoomed in before this feature this will activate. Watch your pencil icon; when you've reached the correct zoom in resolution, it changes from grayed out to active.

Manually De-clicking With the Pencil Icon (Tutorial)

1. Zoom in on your problem area using the Zoom icon (a click will look like a sharp mountain peak).
2. Single click on the pencil icon with the left mouse button. The button will remain down.
3. Move the pencil along the waveform area and depress the left mouse button to make the pencil write a new signal.
4. Many times, just drawing a flat line in place of a click will remove the click and leave no discernable audio residue.

Direct Spectral Editing

(Spectral Editing)



Your Diamond Cut software not only provides you with a wide array of time domain related editing tools, but also several frequency domain tools as well. These are found under the Edit Menu under the rubric of "Direct Spectral Editing" (DSE) or via the Paintbrush icon on the toolbar. It uses the Spectrogram View to present the user with the signal to be edited and editing takes place directly in that view. For full functionality, the DSE requires you to work in Fast Edit mode. In Classic Edit mode, it is only capable of performing the "Replace" function. You can access the DSE feature directly by clicking on the paintbrush icon which will launch the spectrogram as well as the DSE. If you are already in the Spectrogram view, you can also click on the

paintbrush icon to bring up the DSE. The DSE paintbrush allows you to isolate a signal and either amplify it, attenuate it, or interpolate it. The “Replace” feature is very useful for interpolating long-lived events such as a chair moving during a concert or a dropped drumstick or a member of a live audience coughing, yelling or whistling.

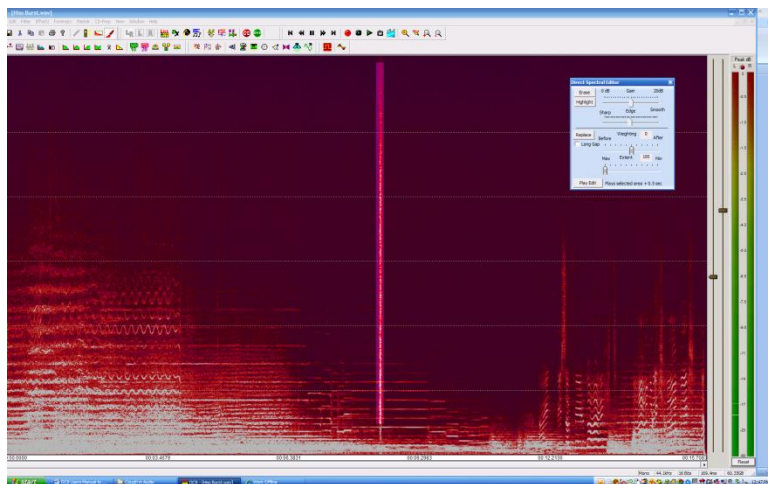


Figure 37 – The Direct Spectral Editor – Highlighting a “Hiss” Event

The primary control for the DSE is the Paintbrush tool. It is used to encircle the signal which is to be processed by drawing a rectangle around it. Horizontal movement of the paintbrush establishes the time range of action and movement along the vertical axis establishes the range of frequencies over which it functions.

The DSE can be operated in any of three modes including the following which can be chosen within the Spectral Editor dialog box:

Erase: Bandstop Filtering (Attenuates the Highlighted Area): This mode provides you with selective filtering capability.

Highlight: Bandpass Filtering (Amplifies the Highlighted Area): This mode is primarily designed to allow the user to focus in on a certain sound in a Forensics type of recording.

Replace: Frequency Domain Interpolation of the Highlighted Area: This mode is not a filter per se, but instead performs a series of calculations to determine what signal most likely should have been in the highlighted area and replaces the original signal with the calculated result.

Three slider controls are also associated with the DSE which are as follows:

Gain: 0 – 20 dB

When operating in “Erase” mode, this determines the amount of attenuation applied to the focused signal; when operating in “Highlight” mode this determines the amount of amplification applied to the signal that you are focused on.

Edge Sharpness: This unit-less slider determines the time and frequency range over which an “Erase” or “Highlight” edit is transitioned. When the control is set all the way towards the right-hand side, that position represents 50% of the size of the bounding frame in both time and frequency. The time edge sharpness is not directly affected by this control; it is calculated automatically.

Replacement Bias: This slider applies to the “Replace” function only and weighs the interpolation routine to the signals distributed about the middle of the interpolated area of the focused signal. When the slider is set towards “before”, the interpolation weighting is more towards the signals preceding the event, and when the slider is set for “after” the interpolator weighs more heavily towards signals that occur past the center of the event of interest. Please note that this control is not in play when performing “Erase” or “Highlight” functions.

Extent Control: This controls how far on each side of the highlighted gap the “Replace” algorithm looks when performing the interpolation. Nominally (by default), it looks roughly 200mS on each side of the gap (when the control is set to 100 - - - all the way to the left side). You can close that down to roughly 20ms at the 10% setting (by moving it towards the right side). Experiment with this control to obtain the optimal replacement. Please note that this control is not in play when performing “Erase” or “Highlight” functions.

Long Gap Checkbox: Normally, the replacement algorithm uses a routine optimized for relatively short gaps, but when the “Long Gap” checkbox is checked, it reverts to a more sophisticated predictive routine. Experiment between the two options in order to obtain the best results.

Right Mouse Buttons: The right mouse button brings up 15 ancillary tools with several of them having particular value when using the DSE including the following four items:

- Edit Spectrograph Properties
- Direct Spectral Editing
- Play From Here
- Undo Last Edit

Of greatest importance is the right mouse “Undo Last Edit” command because the DSE tool is extremely iterative with the user. A lot of trial and error is required in order to find the optimum replacement signal.

Zooming: When you highlight an area with the paintbrush tool, you can zoom into that area using the standard zoom tools (using either icons and/or menus). The system will zoom-in on both time and frequency in a similar manner as the normal time domain zooming function. You can set the maximum zoom-out range using the Edit\Preferences\Spectrograph Properties feature.

Play Edit: This button allows you to play the highlighted area +/- 0.5 seconds so that you can hear the results of the edit in context with the material that surrounds it.

Playing in Context: You can also use the keyboard number keys (1, 2, 3, or 4) to play the edited area of the file in the context of what surrounds it.

Exiting (Escaping) from the DSE mode: Simply depress the Esc key on your keyboard and the system will return to a normal time domain view. (The Esc key is usually located in the upper left hand corner of the keyboard.)

Helpful Hints: You may find it useful to bring up the “Fast Edit History” and the “Time Display” when using the DSE. The Fast Edit History allows you to quickly and easily go back to a previous edit and the Time Display shows you at a glance where you are in your file. Both of these features are found under the “View” menu.

DSE Example:

1. Please set up your Spectrogram as follows:

FFT Size = 4096, Min Hz = 20, Max Hz = 20,000, Freq Axis = Linear, Amplitude Axis = Linear, Window = Blackman, Color Palette = Red-Yellow- Green- Aqua-Blue

2. For DC8, browse to the following Demo file: C:\Program Files\Diamond Cut Productions\DC8\Wavefiles\HissBurstDemo.wav. For Forensics version 8, use the shortcuts placed on the start menu to bring up this demo file.

3. Clone this file so that you do not damage the original and bring up the clone.

4. Listen to the file and note that it has a burst of hiss located at around 8.5 seconds.

5. Under the Forensics Menu, click on the Spectrogram. Adjust the intensity and contrast sliders on the right side of the spectrogram until you can clearly see the “hiss” event (a vertical line which runs from 20 Hz to 20 kHz).

6. Zoom in on the “hiss event”.

7. Click on the Paintbrush and the Direct Spectral Editor dialog box will appear along with the paintbrush which can be moved about using your mouse.

8. Use the Paintbrush/Mouse/Left Mouse Button to circumscribe the “hiss event” with a rectangle running from 20 Hz to 20 kHz and for its entire time interval of occurrence.

9. Click on one of the three modes (Erase, Highlight or Replace) which will cause that particular process to execute.

10. Listen to the result of each of the three modes. After each test run, use the undo function to restore the file to its original state before going on to another test. (Often, the use of the DSE is quite iterative.)

11. Experiment with the various modes and control settings until you become thoroughly familiar with the DSE method of operation.

Note 1: Parameters associated with the Spectrogram like FFT Size and Window do not impact the audio filtering/interpolation associated with the DSE. Those parameters only affect the visual display.

Note 2: Other methods of manual interpolation include Paste Interpolate (Bi-Modal Technique) and Paste Interpolate (Time Domain Technique). Please refer to those sections of this users guide for details.

Mute

This feature uses direct hard disk editing to allow you to mute a selected portion of your .wav file. With Mute, the audio file is reduced to nothing in the selected area, but the area remains. The Mute feature is also useful for getting rid of noise at the beginning or the end of a recording. You can access this feature via the **Ctrl + M** Hotkey or by depressing the right mouse button and using that menu.

Important Note:

Do not mute the beginning or the ending of a .wav file before operating the Impulse Noise filter. Doing so will cause it to function at an extremely slow rate of speed during the muted section, because it will have a very difficult time calculating a signal to noise ratio on a signal containing all zero's. Perform the .wav file muting function after all other filter operations have been completed.

Muting Procedure (Tutorial)

1. Determine the position in the Source or Destination workspace in which you desire to apply the Mute Function and highlight that area.
2. If you change your mind regarding the section that you desire to mute, merely double click the left mouse button anywhere in the workspace area, and the entire file will again become highlighted in yellow. Then, repeat step 1.
3. Click on "Mute" (Edit Menu), and a dialog box will appear which says "Mute will set the selected section of the file to silence. Do you want to continue?" With the left mouse button, click on either "Yes" or "No."

Manual De-Clicking with Mute or Interpolate (Tutorial)

1. Listen to your .wav file using the Play feature, and determine the location of the click, pop, or thud that you desire to eliminate.
2. Zoom-In on the section of the .wav file containing the click using the Zoom In feature.
3. Continue Zooming-In alternately listening to the .wav file until you see the troublesome artifact in the DC8/DC FORENSICS workspace. It will take some training to be able to identify transients visually, so be patient during your learning curve.
4. Using the left mouse button, highlight the transient event, being careful to highlight the complete event, and not just a portion of it.
5. Click on "Edit."
6. Click on "Mute" or use the "Paste/Interpolate" function or depress the "I" (Interpolate) hotkey
7. Zoom back out and listen to the .wav file.

Though you are sometimes actually muting a segment of the audio file, if you use this method carefully, you may not hear it in the final playback. The Interpolate method will generally yield a more transparent result.

Fade In

Fade In does what the name implies when applied to the beginning of a .wav file. You can choose between linear or logarithmic envelopes, and you can also choose the time period for the fade in by selecting the portion of the .wav file over which you desire the fade in to occur. Lastly, you can choose the "start level" for fade in as well as the "stop level." ("Level" is the start and stop loudness for the Fade-In.) Fade In operates on the selected file (which can be the Source or Destination file).

Fade-In Procedure (Tutorial)

1. Listen to the beginning portion of your .wav file and determine the position near the beginning of your .wav file during which you desire to produce a "Fade-In" effect.
2. At the beginning of the sector of the .wav file that you desire to apply the "Fade-In" effect, click down on the left mouse button and keep holding it down as you drag the timing bar (using the mouse) towards the right of the workspace.
3. Stop dragging the mouse when you arrive at a location just prior to the actual beginning of the signal portion of the .wav file.
4. Release the left mouse button. You will notice that the section between the two timing bars will remain highlighted in yellow. This is the sector during which you have chosen to apply a "Fade-In" effect.
5. You can click the right mouse button to hear if you have chosen the correct portion of the .wav file to apply the "Fade-In."
6. Click on the Edit Menu function.
7. Under the Edit menu, Click on "Fade In".
8. Choose the type of Fade that you prefer; either Linear or Logarithmic.
9. Set the "Start Level" slider to zero gain (all the way down). The default setting for this control is zero gain.
10. Set the "Stop Level" slider to 0 dB (unity gain). The default setting for this control is unity gain.
11. Click on OK. The "Fade-In" function will be performed on the chosen portion of the .wav file.

Note: After a Fade-In has been performed; there may be a sector of your .wav file containing some noise at the very beginning just prior to the start of the Fade-In. This can be eliminated with the Mute function.

Fade Out

Fade-out also does what the name implies. It contains all of the features outlined in the "Fade In" description except that it normally works near the end of the file. Fade-Out runs under the Edit menu, and unlike the various Filter functions, operates directly on the selected file (which can be the Source or Destination file).

Fade Out Procedure (Tutorial)

1. Determine the position near the end of your .wav file where you desire to apply the "Fade Out" effect.
2. At the beginning of the sector of the .wav file where you desire to apply the "Fade-Out" effect, click down on the left mouse button and keep holding it down as you drag the timing bar (with the mouse) towards the right of the workspace.
3. Stop dragging the mouse when you arrive at a location in the file where you want total silence to occur.
4. Release the left mouse button. You will notice that the sector between the two timing bars will remain highlighted in yellow. This is the sector during which you have chosen to apply the "Fade-Out" effect.
5. You can click the right mouse button to hear if you have chosen the correct portion of the .wav file to apply the "Fade-Out."
6. Click on the Edit Menu function.
7. Under the Edit menu, Click on Fade Out - - -
8. Choose the type of "Fade Out" which you prefer, either Linear or Logarithmic.
9. Set the "Start Level" slider to 0 dB (unity gain). Unity gain is the default setting for this control.
10. Set the "Stop Level" slider to the zero gain position (all the way down). Zero gain is the default setting for this control.
11. Click on OK. The "Fade-Out" function will be performed on the chosen portion of the .wav file.

Note: After a "Fade-Out" has been performed, there may be a sector of noise after the "fade-out" and the end of your .wav file. This can be eliminated with the Mute function.

Single file Operations

Because of the nature of several operations, the Cut, Copy, Paste, and Fade menu items operate on the file that is currently selected. This means that a Cut will delete a section of the Source file if it is the currently selected file in the workspace. Likewise, a Fade operation will modify the highlighted section of the selected file (Source or Destination).

Snap Selection to Zero Crossing

Remove Editing Glitches with Snap To Zero Crossing

The Snap Selection to Zero Crossing editing feature takes the beginning and ending of a highlighted section of a .wav file and moves both of them to the closest zero crossing point. This is used to minimize the introduction of transients, which could be produced at editing points. This feature is only completely effective on monophonic files since stereo files rarely share zero crossing points on both channels on a highlighted section of a file. Thus, on stereo .wav files, the "Snap" feature moves the highlighted area to the closest average zero crossing value between the two channels. The "Snap Selection to Zero Crossing" feature can be invoked from the Edit menu, via the right mouse button or via the "Q" key on your keyboard. To operate this feature, simply zoom in and highlight the desired section of the file and invoke one of the three options just mentioned above.

Delete All Temp Files

Delete All Temp and Fast Edit Files With One Click Of The Mouse

This command does exactly what the name implies to the temp files in your Temp file (dctmp) directory. Think twice before invoking this command to assure yourself that you are not deleting something important! It also deletes all .pkf (peak) files and .ses (Fast Edit) files.

Gain Change



DC8/DC FORENSICS provides a Gain Change feature that is useful for correcting loudness deficiencies on recordings, or to provide the additional headroom required before running the graphic equalizer filter. Gain Change can be very creatively applied using the contour graphical interface. It can also be utilized globally on a file, or selectively to bring out a weak vocal, etc.

The following is a summary of the control parameters and the range of adjustment provided for the Gain Change algorithm:

- A. Type (Fade In / Fade Out / Gain Change)
- B. Slope (Linear / Logarithmic / Curve)
- C. Gain Ranges:
 - 1. +20/-100 dB
 - 2. +/-20 dB
 - 3. +/-10 dB
 - 4. +/-3 dB
- D. Start Level (dB)
- E. End Level (dB)
- F. Shape (Gain vs. Time):
 - 1. Straight Line (2 Green Cursors) (start and end gain values)
 - 2. Curved Line (4 Green Cursors) (curvilinear inflection point controls)

The Graph shows you how you have programmed the gain to change as a function of the selected .wav file time axis. You can use the mouse to drag the two green control points to establish the time relationship that you desire. Often, a flat line is appropriate; however, sometimes the loudness of old 78s decreased toward the end of the recording by a few dB. This can be corrected by a gain correction starting at 0 dB and ending with perhaps 3 dB (depending on the severity of the problem). The reason this occurred is that the early cutting lathes did not provide automatic gain (or frequency response) compensation controls. When the curve shape is selected, two additional green cursors appear. The two additional green cursors can be moved both vertically and

horizontally allowing you to create numerous curvilinear gain vs. time relationships.

Important Warning:

Digital systems, like analog systems, can be overdriven to the point of "clipping" the signal. This will produce non-desirous distortion (except on rock n' roll). Before applying a gain increase to a .wav file, study the amplitude of the signal and be sure that you are not adding so much gain as to exceed the dynamic range of the system which is most often set to 2^{16} LSB's (or whatever the resolution of the recording that you have digitized). If you do, signals will appear to flatten out horizontally at their peaks on the Source or Destination Workplace displays. If you do indeed notice clipping after a gain change, you can Undo this function.

File Properties

The File Properties feature shows you information concerning the file that is opened in your Diamond Cut software. Figure 33b shows an example of the sort of file header data presented and/or editable.

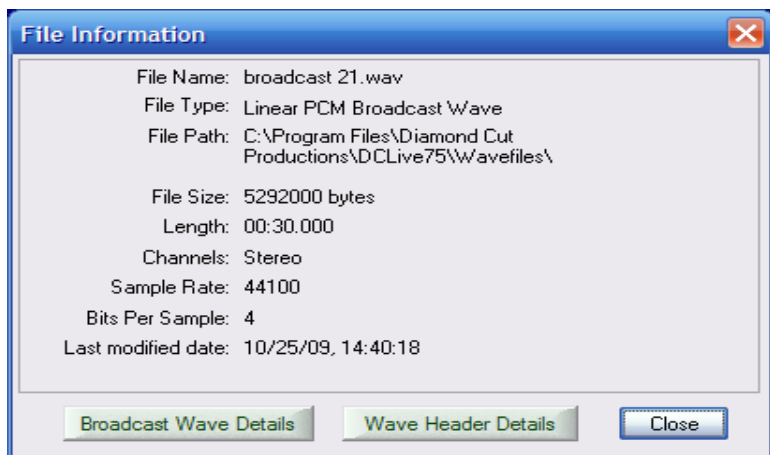


Figure 38 – The File Properties / File Information Dialog Box

Noteworthy is the fact that this information is also available under the View Menu. However, it can only be edited here via File Properties as found under the Edit Menu. If you want to Edit the .wav header, click on the “Wave Header Details” button and a screen similar to this one will appear:

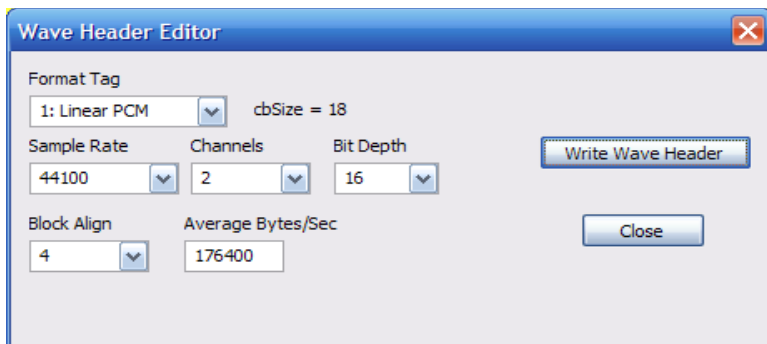


Figure 39 – The Wave Header Editor Dialog Box

From this dialog box, you can edit/change some of the parameters associated with the opened .wav file. In general, it is not desirable to do so unless you are thoroughly familiar with .wav file structures and desire to accomplish some very specific task that could not otherwise be accomplished. After you have changed the desired parameters, click on “Write Wave Header” and the changes will take effect.

Your Diamond Cut Software also supports the Broadcast Wave (BWF) .wav file format. BWF facilitates metafiles in conjunction with .wav files. More information on BWF can be found in the Glossary of Terms section of this users guide. You can use the “Save As” command found in the File Menu to create this sort of a file. If you want to View or Edit a BWF, click on the “Broadcast Wave Details” button and the following Dialog box will appear:

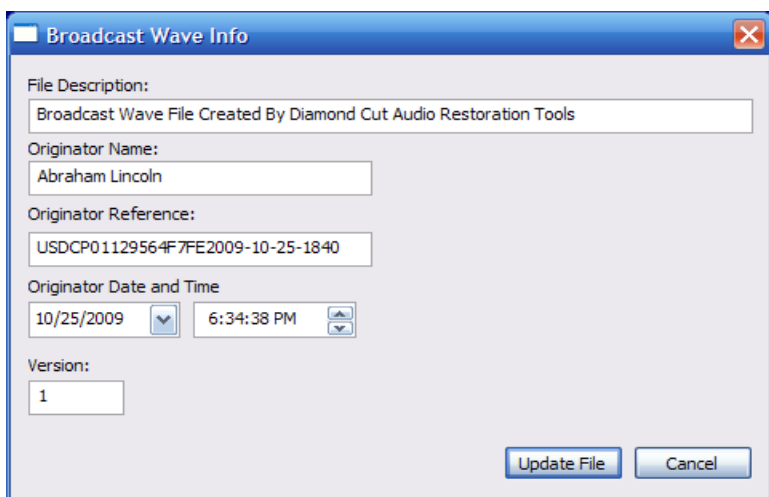


Figure 40 – The Wave Header Editor Dialog Box

As you can see, there are a number of data that you can add to the .wav file header not associated with normal .wav files. After you edit your BWF, click on “Update File” and the file header will be modified.

Playback Controls

If you are new to audio processing on a PC, fear not. The playback controls will be a few familiar friends that you’ll recognize instantly on your tool bar. These tools closely match their “real world” counterparts on every cassette or tape deck that you’ve ever seen. They simply help you quickly and easily navigate through your .wav file in any direction and speed you choose. Because the laws of hardware do not bind us, we’ve been able to improve these controls and make them more accurate and useful than their hardware brethren.

Play



Click here to play your file. If the display is enabled, the highlighted portion shall be played. A shortcut alternative to access this command is via the spacebar.

Scrub Audio



The “Scrub Audio” (scrubber) feature (when enabled) allows you to use your mouse to play a portion of a .wav file with its playback speed being proportional to the left or right position of your mouse. It provides you with a method to find a cue point more easily than via the standard “Play” and “Pause” controls. It is also very useful in file transcription work and also can be used to correct the speed of a file having a variable speed error “on the fly”. To use this feature, just point the mouse where you want to commence play and then click down on the left mouse button. Move the mouse left or right to change the speed of playback. Moving your mouse towards the right advances playback in the forward direction while moving it to the left reverses the direction of playback. Please note that this feature only works with WDM soundcard drivers. It will not work with MME drivers.

Rewind



You click here to rewind the cursor in the file. – This does not operate when previewing a filter.

Pause



This is the key with two vertical lines contained within its perimeter. When activated, the playback of the .wav file will pause at that location. Play can be resumed by activating the play button, depressing the spacebar on your keyboard or by simply hitting the Pause button again.

Fast Forward



You click here to move forward in the file. This does not operate when previewing a filter.

Stop



This is the key with the black square with a green dot contained within its perimeter. It is used to stop either a record or a playback session.

Record



This is the button with the red square. It is used to place the DC8/DC FORENSICS program into Record mode. You can also access this function by hitting the **Ctrl + R** Hotkey. Pressing either the hot keys, menu item or clicking on the Record button will activate the Recording Window. From there, you see the following items:

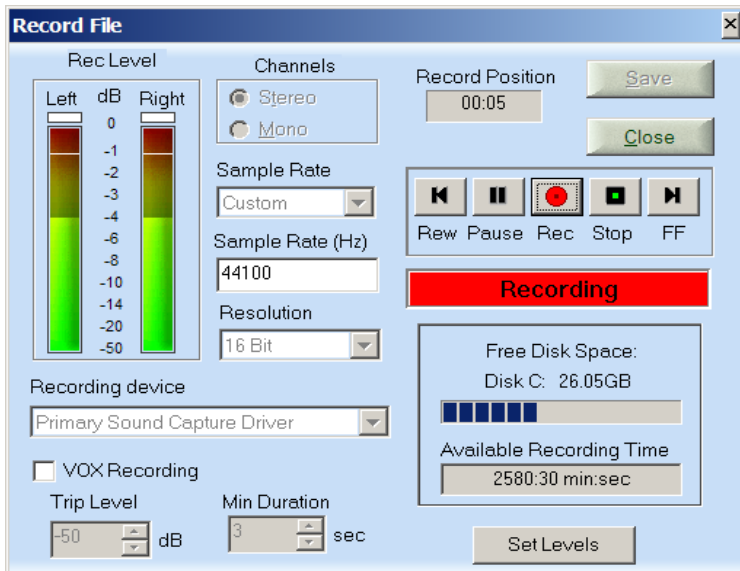


Figure 41 - The Recording Window

- **Recording Level Meters-** Just like on a standard audio or video recorder, you press the Pause button on the Record window, start your source audio playing, and these meters will indicate the amount of signal flowing into your sound card.

No level adjustment can be made here. Simply go to your sound card's mixer to increase or decrease the signal.

- **Recording Device-** If you've set up your sound card(s) correctly, this list will display them and allow you to choose your desired recording device. If nothing is displayed here, it's time to revisit both Windows and your sound card installation instructions. The recording device setup is also available under the Edit/Preferences/Soundcard pathway.
- **VOX Recording-** This check box activates VOX recording. If you want to trigger your recorder automatically on some sort of input signal, this is where it happens. You simply set a Trip Level that is measured in dB. Record will remain paused until a signal above the level you've indicated is received on input. Record will stop when the system senses no input at that level for a period determined by the "Minimum Duration" settings. For more info, see below.

Trip Level- Threshold Range is from +1 dB to -100 dB. This "trips" the system into recording when the desired level is reached.

Minimum Duration- Time Range is from 0 to 360 seconds. After the system begins to record, it will stop automatically when the desired signal stops occurring for the amount of time you've dictated in this field.

- **Channels-** Allows you to select Mono (1 channel) or Stereo (2 channel) recording
- **Sample Rate-** Allows you to choose your desired sampling rate. Here are the choices:
 - a. 192.00 or 96.00 kHz- Normally associated with DVD recording and playback, these rates provide bandwidth well beyond the audio spectrum.
 - b. 48.00 kHz- This rate will produce slightly higher values of bandwidth than 44.1 kHz, but is primarily used for pro-audio applications.
 - c. 44.10 kHz- This rate will produce a full 20 kHz recording bandwidth, however it consumes disk space at a fast rate - - - 5.29 Mbytes per minute per input channel. This is the same sampling rate used on commercial Compact Discs.
 - d. 22.05 kHz- This rate will produce a recording bandwidth of 10 KHz, which is adequate for the

restoration of old acoustical recordings. This setting consumes disk space at the rate of 2.645 Mbytes per minute per input channel.

- e. 11.25 kHz- (This rate will produce a recording bandwidth of only 5 kHz, which is useful for the restoration of old spoken word recordings, telephone conversations, and first generation movie soundtracks.) It consumes disk space at the lowest rate of only 1.3225 Mbytes per minute per input channel.
 - f. Custom- Using this setting, you can fill in any sampling rate you choose under the Custom “Sample Rate (Hz)” field. This could be useful in some situations; provided your sound card can support anything but the standard sample rates. The range of sample rates which can be entered as a Custom value is any integer in the range extending from 100 Hz up to 192,000 Hz.
- **Sample Rate (Hz)**- If you choose “Custom” sampling rate, this is where you manually type in the sampling rate you want. (Your sound card must support this rate, however in order for it to work properly.)
 - **Resolution**- This is where you’ll choose the bit width of your recording. You can choose from the following:
 - a. 8 bit (Games)
 - b. 16 bit (most common - CD Quality Audio)
 - c. 20 bit (professional audio)
 - d. 24 bit (professional audio)
 - e. 32 bit Float (Floating Point for special applications)
 - f. 32 bit Int (Integer for special applications)
 - **Record Position**- This window, like your tape counter, simply tells you where you are in your recording.
 - **Transport controls**- Just like a common tape recorder, you can Rewind, Pause, Record, Stop, and Fast Forward in your recording.

- **Current System Status-** This gray box normally defaults to the “Stopped” position, but also turns red on Record and blue on VOX Waiting.
- **Free Disk Space-** Tells you how much space is available for recording on your hard disk.
- **Available Recording Time-** Does a quick calculation on the fly of your hard disk space, sampling rate, bit width and gives you an estimate of how much recording time remains.
- **Spacebar Control-** Normally, the spacebar controls audio playback. However, when the record dialog box is showing, the spacebar reverts to a toggle function alternating between record and record-pause mode of operation.

VOX Recording

As indicated above in the Record window, VOX (Voice activated Transmit), or signal activated recording, is possible with DC8/DC FORENSICS. This feature allows the recorder to start and stop itself based on the presence or lack of presence on an audio input signal. The recording activation process is triggered by signal level sensing as determined by your setting of the Trip Level control. The record de-activation process is triggered by a combination of the signal level sensing system in conjunction with the Minimum Duration control setting. For example, you will want to set the Minimum Duration to a value long enough so that the recorder does not stop between songs on an LP. In that example, you would probably want to set the control for about 10 seconds. To use the VOX feature, bring up the Recording window and check VOX recording and set the Trip Level and Minimum Duration appropriately for your specific application.

In the DC Forensics Version, you can also perform automatic time stamping of VOX events when you have Log to Disk selected (see the Live Mode section for more information). This function adds a time stamp for each recording event and is accurate to less than 1 second.

Extended Recording

Record Material of any Length- No 2 Gbyte file Limit

Wave files are limited in length to 2 Gbytes. In some situations, it is desirable to record for a period of that exceeds this 2 Gbyte limit. The Extended Recording Feature accommodates this situation. It works by automatically opening a new file when a particular file approaches the 2 Gbyte limit. When you save the Extended file(s), you will see that the file(s) will be named in the following format:

xxx_part1
xxx_part2
xxx_part3
xxx_partn

Each file will have to be played or processed independently thereafter.

Recording Signals onto your Hard Drive (Tutorial)

1. Before your first recording, please review the quick tutorials for Getting Started.
2. This process needs only to be performed once, if you only have one sound card. Your chosen setting will be saved. However, if you have multiple sound cards, this process will have to be repeated anytime you desire to change the input or output configuration to DC8/DC FORENSICS.
3. Now, click on the red Record button found on the toolbar. You will see the above Record window. Determine your settings and select the desired recording device from the drop down window.
4. Click on the Pause (Record Pause) button in the recording window. If your source is analog, adjust the output level of the sound source or the input sensitivity of your sound card until the green VU meter bar graphs are modulating vertically to the maximum degree possible short of activating the red overload indicators. Please note that this adjustment may consist of a hardware control (gain or volume) of the output signal feeding into your sound card OR you may use the input mixer of your sound card to perform this function. Sample the loudest portion of the sound source to assure that no

overloading will occur when the transfer is made to your hard drive. Please note that digital sources are not gain adjustable. Whatever the gain settings that were used initially on this type of source, translates to the gain that you will get when transferring to the hard drive via DC8/DC FORENSICS. If the original analog to digital transfer was either overloaded or under recorded, it will remain that way.

5. To commence recording, simply click on the Record button.
6. The "Record Position" is analogous to the tape counter of a conventional tape recorder. It uses real-time measurement. (Minutes: Seconds)
7. To pause the recording, click on the Rec. Pause button on the toolbar. You may continue to record from the pause position by repeating the method outlined in step #5.
8. Recording can be terminated by clicking on the Stop button. (Stop is the square button containing a smaller black square within its perimeter).
9. To save your recording, click on the Save button in the "Record File" dialog box. You will notice that name has already been assigned to your file. You can either keep the assigned name for your file, or rename it at this time.

Important Note:

DC8/DC FORENSICS is compatible with .wav files that were originally created by other programs. It is not necessary to record a .wav file using the Diamond Cut recording routine in order to use its processing features.

Play



This is the key with a black and green arrow, which is pointing towards the right. This key is used to play your file. The playback will begin at the leftmost portion of the yellow highlight area of your .wav file. You can also access this function by pressing your keyboard spacebar.

Playing a Wave file (Tutorial)

1. Launch DC8/DC FORENSICS.
2. If you haven't already done so, you must define your Output Device.

- A. To do so, click on “Edit /Preferences/Soundcard and then “Device I/O Selection”.
 - B. Choose the output device that you desire, and then click on OK.
3. Next, click on “File”, and then on “Open” and select the file you wish to play.
4. You will notice that the waveform will appear in black on a yellow background. This area of the window is the Source workspace. If your .wav file is monophonic, you will see just one waveform displayed in the workspace. If your .wav file is stereophonic, then you will see two waveforms displayed, with the top waveform representing the left channel and the bottom waveform representing the right channel. The length of your .wav file is displayed in Minutes: Seconds: Milliseconds in the timeline at the bottom of the Source Workspace. It’s a good idea to also run on the Time Display window by clicking on View and then checking Time Display.
5. To play the file, click on the Play button on the toolbar or depress your keyboard spacebar. You will notice the playback cursor begin to "march" across the workspace as the system plays the file.
6. To terminate playback, click on the stop button on the toolbar or hit the space bar.
7. If only a portion of the .wav file had been highlighted for playback, and now you desire to playback the entire file, double click on the workspace background, and the entire file will become highlighted, and ready for playback.

Pausing and Resuming Playback

1. To pause playback, click on the Pause button on the toolbar
2. To resume playback, you can either click on the Play button, Pause button again or depress the keyboard spacebar.

Playing and Pausing using the Right Mouse Button

1. To start the playback of your .wav file at a specific location, use your mouse to move the mouse arrow pointer to the desired location in the Source or Destination workspace.
2. Single click the right mouse button and you will notice a "Play from here" option in the menu. Click it.

3. The file will continue playing until its end unless you click the right-hand mouse button again anywhere in the Source or Destination workspace. Playback will terminate the moment the right-hand mouse button is clicked.

Play Looped



This button allows you to play either the entire file or a highlighted area over and over again back to back. Playback will continue until the button is depressed again or the space bar is pressed. You can also access this feature by using the “L” Hotkey.

Note: This feature will not work if the “Play From Cursor” method of playback is selected in the Preferences menu.

Time Bracketed Play Range

This feature allows you to play a range of time around a selected area. It is useful when you need to listen for small anomalies on a sound recording in order to find their exact location. You can choose between the following, which can be activated from the keyboard using the following keystrokes (shown in brackets): To use this function, select an area by clicking and dragging. You can play this selected area by using the spacebar or the Play button. You can also play the selected area PLUS a specific amount of time in front and behind the selected area by simply hitting the 1 thru 4 keys on your keyboard. For example, if you select an area 10 seconds in length, hitting the 2 key will play starting at a point 2 seconds before the selected area and continue for an area representing 2 seconds after the selected area.

Play Range +/- 0.5 seconds (1)

Play Range +/- 1 second (2)

Play Range +/- 2 seconds (3)

Play Range +/- 4 seconds (4)

Timer Recording

The new Timer Recorder feature is useful for such applications as recording your favorite radio broadcast at a defined time, or to

commence a surveillance recording based on a particular time of day. It is a single event 24-hour timer, which clears itself after the event has passed, or can repeat its action the following day by checking the “Re-Occurring Event” box. To set a “Start Time”, click on the number to be changed in the “Start Time” field, and use the up and down arrows until the readout agrees with the desired start time. The same process is used for setting up the timers “End Time.” The Re-Occurring Event checkbox will cause the recorder to record the same time span every day. Use the “File Name for Recording” in conjunction with the “Browse” feature to define the file location and name into which the recording shall be placed. ***You must leave the software running for the Timer Recorder feature to work and you must close the Timer Recording window.*** You can minimize it, but do not exit the program. If you want to disable the Timer, click on the “Cancel Timer” checkbox.

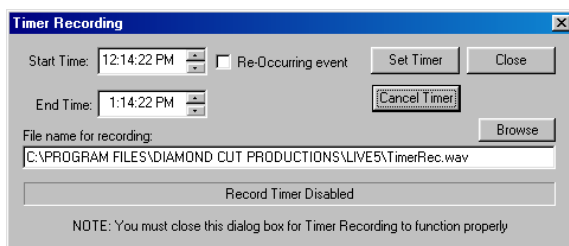


Figure 42- The Timer Recording Window

Click on the red record button found on the tool bar in order to set your recording parameters such as sample rate, resolution (bit depth), channels (stereo or mono), and sound card location. The record dialog box will appear on your screen. Then, select the parameters that you desire and close the recorder. Those values will be saved and used by the Timer Recording feature.

Make Waves Signal Generator

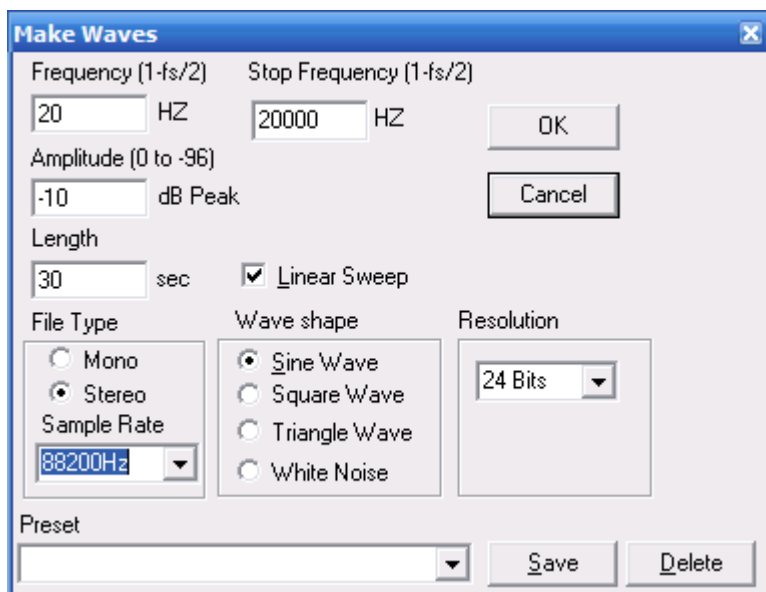


Figure 43 – The Make Waves Signal Generator

The “Make Waves” signal generator feature provides you with the comprehensive capabilities associated with a fully programmable audio signal generator (sometimes referred to as a function generator by engineers). It can produce Sine Waves, Square Waves, and Triangle Waves of adjustable frequency and amplitude. The frequency of these waveforms can also be swept from one value of to another. The Make Waves generator is also capable of producing random waveforms of the White Noise variety. (Pink Noise, Brown Noise or Seismic Noise can also be produced when random noise is used as a signal source in conjunction with the appropriate Multifilter or Paragraphic EQ conversion preset.) Make Waves is useful for calibrating and verifying the performance of the audio equipment used in your sound restoration laboratory, especially when used in conjunction with the onboard Spectrum analyzer. These tools will also be useful to help you better understand the functionality of some of the filters provided in the application. The Sweep and Random generator is especially useful for

characterizing the frequency response of electrical and acoustical systems. The Triangle Wave generator is useful for characterizing the time-domain linearity of an audio system or component. Included with the Make Waves Generator presets are all of the musical notes running from C0 to D9# (referenced to a 440 A4 of the equal-tempered scale) as well as a number of useful test signals from which to choose. The following controls with their adjustment range are provided:

1. Start Frequency: 0.01 Hz to 100,000 Hz.*
2. Stop Frequency: 0.01 Hz to 100,000 Hz.*
3. Length: (Duration of the tone burst) 10 Millisecond to 600 seconds. (Data entry is in seconds.)
4. Amplitude: 0 dB to -145 dB
5. Linear Sweep check box (on or off)
6. Sine, Square, Triangle Wave or Random (white noise)* selector
7. File Type (Stereo or Mono)
8. Sampling Rate {Factory Pre-Programmed} (8, 11.025, 22.05, 44.1, 48, 88.2, 96, 176.4 and 192 kHz)
9. Sampling Rate {User Programmable} (100 Hz to 210 kHz)**
10. Resolution (8, 16, 20, and 24 bits)
11. Presets: Includes common signals as well as the Musical Scale ranging from C0 (16.35 Hz) to D9# (9,956.06 Hz)

*Note 1: The frequency programming precision of the Make Waves Generator in the DC Forensics Version is greater than the standard DC8 product. Keyboard entry of the desired frequency must be used to attain this higher level of resolution. The actual frequency resolution in the DC Forensics Make Waves Generator is 0.01 Hz (10 mHz or millihertz). Thus, values like 59.99 Hz or 60.01 Hz are permissible.

**Note 2: To create a Make Waves file having a custom sample rate value up to 210 kHz, simply highlight the sample rate field and use your keyboard to enter the desired sample rate value.

The following chart summarizes the Harmonic Distortion components produced by the various waveforms provided by the "Make Waves" generator:

Wave-form	THD %	3 rd Harm	5 th Harm	7 th Harm	9 th Harm
Sine	0	0	0	0	0
Square	44	33.3	20	13.8	10.8
Triangle	12	11	4.1	2.0	1.3

***Note:**

Using the Making Waves Signal Generator (Tutorial)

1. Click on the "Edit" Menu with the left mouse button.
2. Click on "Make Waves" with the left mouse button.
3. Choose between Sine Waves, Square Waves or Random by clicking on the appropriate box with the left mouse button.
4. Set the desired "Frequency" by clicking on the number(s) that you desire to change with the left mouse button. Use the keyboard to enter the new values with the "Start frequency" slider control. The allowable range is from 0.01 Hz to 100,000 Hz. (100 kHz)
5. Set the "Length" of the file, which you desire to create using the mouse, and direct keyboard entry. Data entry must be in terms of seconds. If you desire 2 minutes, enter 120. If you desire 10 Milliseconds, enter 0.01.
6. Set the "Amplitude" which you desire, anywhere from 0 dB to -145 dB. 0 dB is the largest signal value which you can produce, with the peak value of the waveform being + / - 32,767 LSB's referenced to a 16 bit file.
7. Click on "OK" and a Source file will be created for you containing the signal that you have just defined.

Important Note 1:

If a sweep generator function is desired, click on "linear sweep" and then adjust the "stop frequency" control to the desired value. The generator will then produce a linear sweep of frequencies ranging from the start frequency value to the stop frequency value over the interval of time defined by the "length" control. The Sweep and Random

Generator is useful for characterizing the frequency response of electrical and acoustical systems.

Important Note 2:

Pink noise can be created from the random white noise generator by applying the “White to Pink Noise Converter, 20 kHz” found in the preset list under the Multifilter. First, create a sample of random (white) noise using the Make Waves Generator. Use a sampling rate of 44.1 kHz for this procedure. Next process the resulting file through the indicated Multifilter preset and the resulting file will be Pink Noise.

Important Note 3:

Brown and/or Seismic Noise can be created in a similar manner as described in Important Note 2. However, the converters are found as presets under the appropriate names in the Multi-Filter.

Important Note 4:

A number of factory presets are provided for the Make Waves Generator consisting of commonly used audio signals. You can add to that list as desired.

Change Sample Rate / Resolution

The Change Sample Rate feature allows you to convert a .wav file from any common sample rate to any other common sample rate. It also allows you to convert from any common resolution (or file depth) to any other common value. These features are provided with a user selectable interpolation quality since improved interpolation takes more CPU horsepower and therefore more time to make the conversions. This way, the user can make this tradeoff. The following controls are provided with the associated ranges of the Change Sample Rate / Change Resolution Feature:

New Sample Rate: 11,025, 22,050, 32,100, 44,100, 48,000, 88,200, 96,000, 192,000 Samples/sec.

Resolution: 8 Bits, 16 Bits, 20 Bits, 24 Bits, 32 Bits FP

Conversion Quality: Choose between –

- CD (16 bit interpolation)

- Pro (24 bit interpolation)
- Master (32 bit interpolation)

Dither: Choose between –

- None
- Flat Spectrum: White noise is used as the dither. This is usually not the best choice unless you are doing analytical work.
- Triangular High Pass: A shaped noise spectrum that does a good job of reducing distortion. The name comes from the type of digital filter used. The noise is biased to the high frequency end of the audio spectrum and is less audible than the flat spectrum.
- Noise Shape 2: Our own proprietary noise shaped spectrum. This distribution has less midband noise than the triangular spectrum, but more at the high end of the audio band.

Note 1: In general, choose Triangular or Noise Shape 2 with your ears making the decision as to what is best.

Note 2: Usually dithering is used when lowering the bit depth...like going from 24 to 16 bits.

To perform a conversion, bring up the file to be converted in the source window. Select the desired conversion parameters from the outlined choices. Run the filter and the results will be displayed in the Source window after a period of processing time. The file will be re-named by incrementing its numerical extension by a value of one, so long as there are no other files with that same incremented name in existence in the chosen file directory.

Preferences

Preferences allows you to customize the system up to your own liking.

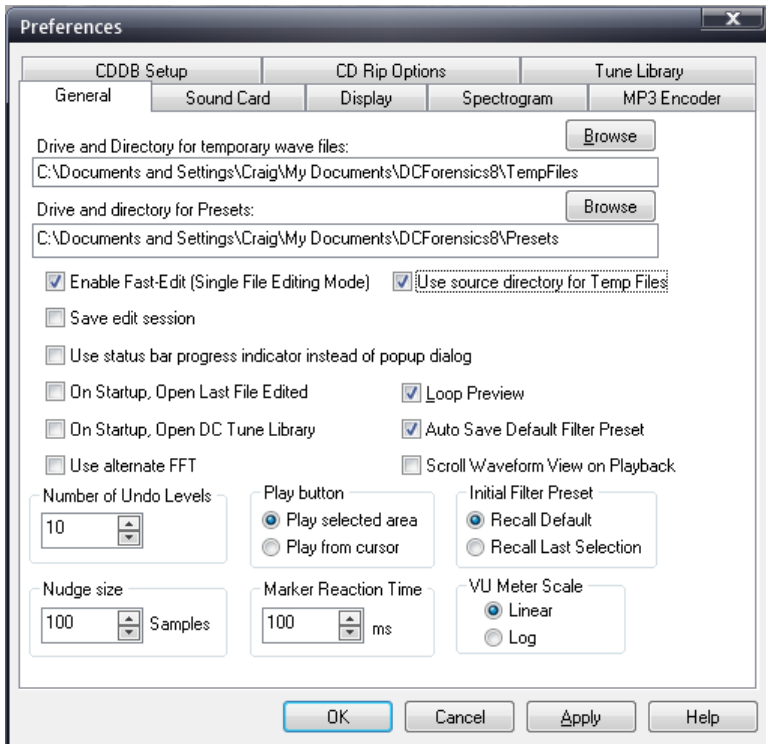


Figure 44 - Configure the System Your Way!

Preferences are broken down into eight sub-categories that can be accessed by clicking on the desired Tab at the top of the Preferences window.

General:

1. Drive & Directory for Temporary .wav files (999 maximum)
2. Drive and Directory for Presets
3. Enable Fast Edit Check box- This turns Fast Edit Mode on or off. If a file is open and you want to change editing modes, your file must be closed before or after switching or you must

close down the program and restart it in order for this change to take effect. When the Fast Edit mode is turned off, you will be operating in the Classic Edit mode.

4. Save Edit Session– When this is checked, you will be queried each time you exit the program as to whether or not you want to save the File Edit History file. If this feature is not checked, the Edit File History file will be automatically deleted when you exit the program. By saving the edit history, you maintain the ability to go back to any previous editing state without destroying the original file. Keep in mind that this action will use more disk space since all of the intermediate edits are stored in temporary files on your hard drive.

Note: *The Edit History (or Session File) is stored in the same directory as the .wav file. It contains the file extension .ses and should not be deleted manually because it will leave the associated temporary files on your hard drive.*

5. Use Status Bar Progress indicator rather than popup dialog- When checked, the progress of a particular filter will be indicated at the bottom of the screen using a progress bar rather than the classic popup dialog box.
6. Auto Save Default Filter Presets- This feature allows you to choose between the action of returning the filter setting called Default to factory values or to your last personal setting after closing a filter.
7. Number of Undo Levels Saved- This parameter allows you to choose the number of undo levels of destructive editing which DC8/DC FORENSICS will maintain as stored files on your hard drive. The default value for this parameter is 10.
8. Nudge Size- This parameter defines the resolution of the left and right arrow keys on your keyboard as they apply to the .wav file-highlighting feature of DC8/DC FORENSICS. This parameter is defined in terms of samples. After highlighting a portion of a .wav file, you can fine tune or “nudge” the highlighted area using the left and right arrow keys, and the Shift key. The resolution of each click on an arrow key is defined by the value of “nudge size.”
9. Play Options- Allows you to choose “Play Selected Area”, which defaults playback to the beginning of a selected area of

the beginning of the file each time you press “Play”, or “Play from Cursor”, which, like a tape player, starts from wherever you last played the file.

10. Marker Reaction Time- This parameter allows you to compensate for you and your computer system’s lagging reaction time when dropping markers on the fly using the “M” keyboard accelerator. Its units are calibrated in milliseconds.
11. Loop Preview on or off- This feature, when checked, will cause a “previewed” section of a .wav file to repeat itself endlessly until the filter or effect is canceled or the preview button is clicked on a second time.
12. VU Meter Scale- Choose from Linear or Logarithmic scales.
13. “On Startup, Open Last File Edited” is an alternative start-up mode that you can choose by checking this box.
14. Scroll Waveform View On Playback: Normally, the play cursor moves across the time domain display as a .wav file is being played and stops at the end of the highlighted area. “Scroll Waveform View On Playback” is an option wherein the display will keep pace with the playback position. As the cursor reaches the end of the display area, the display will page one frame to the right and the cursor will start again at the left hand side. The Zoom level is maintained and can be changed with the Zoom X2 feature while playing. You will need to use the “Play from Here” function found on your right mouse button to use this feature or use the Zoom X2 functions to display an area that is different than the highlighted area.
15. On Startup, Open DCTune Library: When checked and the software is launched, the DCTune Library will automatically be launched too.
16. Use Source Directory for Temp Files: This mode is particularly useful for Forensics users. It allows all of your work to be put into a single directory for easy archiving. When this mode is not used, it may be difficult to identify all of the files used in a fast edit session because they will be in the temp directory. This is only available in the Forensics version.
17. Use Alternate FFT: This feature is a diagnostic tool used by the Diamond Cut Company.

Sound Card:

1. **Driver Type Check Box:** Select between MME and WDM, depending on which driver best works with your particular sound card. If your sound card is older, chances are that the MME selection will work better. If you have a newer sound card, then WDM may be your choice. We recommend always trying WDM drivers first.
2. **Preview Buffers (2 to 50)** (raise this value if you encounter stuttering during preview)- This parameter applies to preview mode only. It allows you to choose the size of the buffer space that is used by the preview feature. You can select between 2 to 50 buffers. 1 buffer = 4096 samples. The larger the buffer which you choose, the longer the sample which you will hear before the system repeats itself (stutters) if your system is not fast enough to run a particular algorithm in real time. However, the larger the buffer, the longer will be the delay time before you hear the results of a preview session. As a general rule, low sampling rates coupled with low bit depths (low resolution) require smaller buffer sizes compared to deep bit depths (high resolution) and high sampling rates. Also, .wav files that are stereophonic require more buffers than monophonic recordings.
3. **Input Device Selector** (Browse to desired sound card Input)
4. **Output Device Selector** (Browse to desired sound card Output)
5. **“Re-Initialize on Play”** sometimes fixes problems with WDM soundcard drivers, but is not applicable to MME’s. If you are observing irrational behavior with your WDM drivers, try checking off this option as it may clear up your soundcard problem.
6. **Play Chimes after Long Operation Completes** Checkbox – Provides you with a distinctive audible enunciator after the completion of long operations such as the Batch File Editor, CD Ripper, CD Burner, and the Data Disk Burner.

Important Note: Most tech support calls for “no playback” or “no record” symptoms relate to sound card issues. DC8/DC FORENSICS uses all standard Windows calls to playback and record audio and does not talk to your specific sound card directly. Therefore, any sound card that works under Windows will automatically work with DC8/DC FORENSICS. If you are having problem playing or recording audio, make sure your sound card is working with other applications and make sure you have the correct inputs and outputs selected under Edit/Preference/Soundcard.

Display:

1. Check “Don’t use peak files (.pkf) for display” (no time display shown when checked). *This does not apply in Fast Edit Mode.*
2. Check “Sync Mode Scroll Tracking” (synchronizes source and destination displays)
3. Check “Clean Display” which removes .wav file path name from the display
4. Check “Show Splash Screen on Startup” (saves time following software launch)
5. Display Colors- This set of parameters allows you to choose the Source and Destination Workspace background, highlighted, and waveform colors. When you click on either parameter, a color palate will appear, and you can click on the color combinations that suit your vision the best.
6. Display Time Format:
 - a. Minutes: Seconds
 - b. Samples
7. Display X-Axis Time Scale
8. Display Font Size (enter value in terms of “points”, 2 to 16)

Spectrograph:

Note 1: Please refer to the Spectrograph section of this manual for details.

Note 2: Left mouse double clicking in the Spectrograph window can also access the Spectrograph preferences.

mp3 Encoder

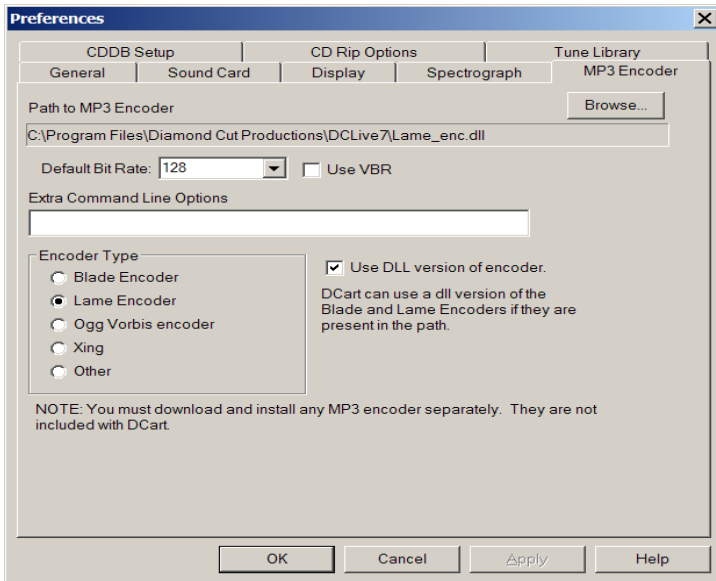


Figure 45 - mp3 Encoder Preferences Setup

Setup is provided for an externally provided MP3 Encoder. At this time, we accept popular 3rd party encoders that are available from the Internet. To allow the program to Save or Batch Process files into MP3 format, you must download an encoder and choose it in this screen. The Browse button allows you to show DC8/DC FORENSICS where the encoder is located on your hard drive. In general, higher bit rates produce better sonic results at the expense of greater memory consumption. Note that the system supports both CBR (Constant Bit Rate) and VBR (Variable Bit Rate) encoding techniques.

Tune Library:

Please refer to the DC Tune Library section of this documentation for detail on these preferences.

CDDB Setup:

Your software has the capability to “Rip” Red Book CD Audio to .wav files. For details regarding the operation of this feature, please refer to the section of the manual called “Rip CD Tracks” which is found under the Filter Menu of the program. The CDDB Setup allows you to establish the parameters so that your system can fetch Track titles off of the Internet and use them as Track names rather than Track numbers. Here is a screen shot of the CDDB Setup tab:

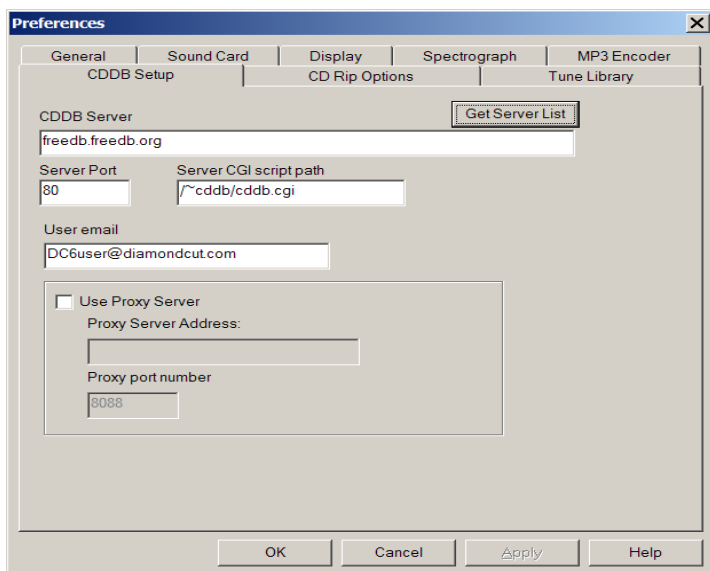


Figure 46 - CD Database Setup

You will need to be on the Internet for the CDDB feature to function. Generally, the default settings provided will work for you. However, it is a good idea to click on the “Get Server List” and select the server closest to your locale. If you are running a firewall on your system, you will have to check the “Use Proxy Server” box and enter the proxy server address and proxy port number.

File Split and Recombine

Sometimes it may be desirable to split a stereo .wav file into its left and right components, establishing individual .wav files for separate channel processing. For example, if one of the stereo channels needs some equalization but not the other, using this splitting and re-combining tool can facilitate that function. This easy to use tool allows you to either Split a Stereo file into two Mono files or combine two Mono files into one Stereo file. You can select whether or not to open these files after the process is complete. Note that the L/R buttons on the toolbar allow instant selection of either or both channels for playing and processing.

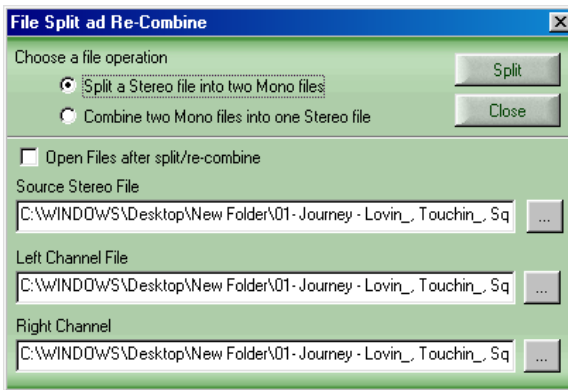


Figure 47 - Easily Split and Recombine files with this feature

Manage Presets

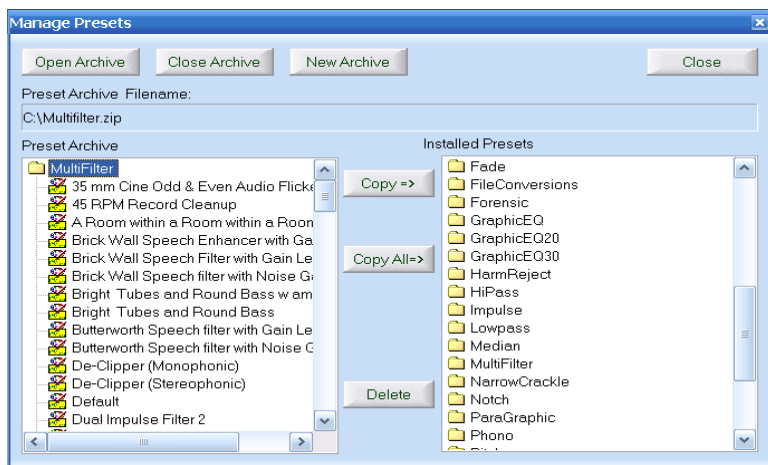


Figure 48- The Preset Manager

The Presets Archive stores all presets, either factory produced or user produced in folders according to their filter. A new feature of DC8/DC FORENSICS is the ability to share presets you have created with other users via email or web pages. You can export any or all of your presets and share them with other users by sending them a single file. See www.diamondcut.com. Click on the Users Forum and you will see the preset exchange area. This exchange provides both uploading and downloading capability with an area for you to describe the preset(s) that you are posting.

Current presets are stored in a Presets folder in the Diamond Cut directory on your hard disk as a single small file for each preset. Since DC8/DC FORENSICS comes with over 1000 presets, it would be cumbersome to find and share them by sending or receiving one file for each preset. The Manage Presets window makes this easy. You simply create an Archive by clicking on the button and giving the new archive a name. Now you can copy any individual or group of presets into this archive. Double clicking a preset folder will open it and reveal its individual presets. This archive file will exist as a single filename .zip file on your hard disk and will contain all the presets you copied into it.

Copying presets does not delete them from your own preset list. This single .Zip file can then be sent to another user and opened using the Open Archive function. The user can then place any or all of the received presets into any of his or her personal preset folders.

Saving a Preset (Tutorial)

1. Establish the desired filter settings and states for a particular filter application.
2. Click on the "Save" button.
3. Using your mouse, place the cursor at the beginning of the data entry field, and double click the left mouse button.
4. Delete any characters in the data field with the "delete" key on your keyboard.
5. Type in a descriptive name for your setting (up to 32 characters in length).
6. Click on "OK". Your setting will then have been saved.

Recalling a Preset (Tutorial)

1. With the left mouse button, click on the down arrow located on the right hand side of the setting list (located at the bottom of the Filter Dialog Box).
2. With the left mouse button, single click on the filter setting preset description you desire from the listing.

Deleting a Preset (Tutorial)

1. With the left mouse button, click on the down arrow located on the right hand side of the setting list (located at the bottom of the Filter Dialog Box).
2. With the left mouse button single click on the filter setting preset that you desire to delete.
3. With the left mouse button click on the "Delete" button. A question box will appear. If you still desire to delete the particular filter preset, click on "yes." If you do not, click on "no."

The Filter Menu

Batch File Editor

A Batch File editor is provided, so that you can assemble a group of .wav files with similar problems and apply filters and their associated parameters to this entire list of files. This will allow you to create your setups and then go out and mow the lawn or perform other chores while your computer and DC8/DC FORENSICS are performing their chores unattended by you. The number of files that the batch processor is capable of handling results from a total path length of 128K characters as the sum total for all file paths that you want to batch process. The maximum number of characters allowed per file name and path is 2048. The system will remember the last path that you used.

The following items are supported by the Batch File Editor:

- 10 Band Graphic EQ
- 20 Band Graphic EQ
- 30 Band Graphic EQ
- Adaptive Filter
- Auto Leveling System
- Averaging Filter
- Band Pass Filter
- Big Click Filter
- Brick Wall Filter(s)
- Change Sample Rate or Resolution
- Change Speed
- Concatenate Files
- Continuous Noise Filter
- Convert AIFF Files to Wave Files
- Convert mp3 Files to Wave Files
- Convert Wave Files to AIFF Files
- Convert Wave Files to mp3 Files
- De-Clipper
- Big Click Filter
- DirectX Plug-ins
- Dynamic Spectrum Subtraction Filter (DSS)

- Dynamic Noise Filter
- Echo Effect
- Expert Impulse Noise Filter
- EZ Clean Filter
- EZ Enhancer
- EZ Impulse Filter
- EZ Forensics Filter
- File Conversion Feature
- Filter Sweeper Filter
- Harmonic Reject Filter
- High Pass Filter
- Low Pass Filter
- Median Filter
- Multi-Filter
- Narrow Crackle Filter
- Normalize
- Notch Filter
- Overtone Synthesizer
- Paragraphic EQ
- Phono Preamp (VPA)
- Polynomial Filter
- Punch and Crunch Effect
- Reverb Effect
- Spectral Filter
- Stretch and Squish Effect
- Sub-harmonic Synthesizer
- Virtual Valve Amplifier (VVA)
- Cell Phone Noise Filter
- Voice Garbler
- Auto Voice Filter

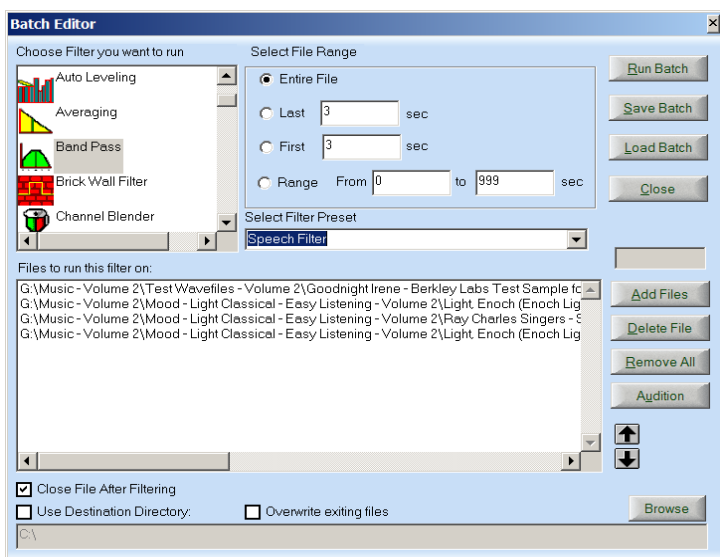


Figure 49 - Process Multiple Files Simultaneously!

The Batch File editor includes the following controls:

1. A "Filter Menu" which contains all of the DC8/DC FORENSICS filters.
2. A "File Time Range Selector" which allows you to select the time interval of the .wav files on which you desire to apply the filter / filters. You can select between
 - A. Entire File
 - B. Last XX Seconds
 - C. First YY Seconds
 - D. Range from XX to YY Seconds
3. The .wav file listing which you will create
4. A "Select Filter Preset" box
5. "Run Batch" button
6. "Save Batch" button (for saving your batch settings)
7. "Load Batch" button (for recalling a batch)
8. "Close" button
9. "Add Files" button to bring up the file manager
10. "Delete Files" button

Auto Leveling

Match Volumes of Numerous Wave Files with Auto Leveling

The Auto Leveling routine is contained only in the Batch Editor which is found under the Filter Menu. It allows you to make .wav files sound very similar in loudness across a number of files. It performs this function by calculating a combination of the RMS and Peak values of each of the files and normalizing them while accounting for both parameters and assuring that none of the .wav files overload. This functionality is quite different in functionality from the Gain Normalize feature in which a single file is normalized in gain with respect to its own peak value. The Auto Leveling routine has no specific controls unto itself. It uses the generic controls associated with the Batch Editor. To operate the Auto Leveling feature, highlight the Auto Leveling Icon. Then merely use the “Add Files” feature in the Batch Editor to list the files of interest by placing them in the “Files to Run this Filter On” listing box. Lastly, click on the “Run Batch” button. The resultant processed .wav files will be found to be very similar in overall loudness.

Concatenate Files

Concatenate Files Feature Available in the Batch Processor

Another feature, which is only available in the Batch File Editor, is the Concatenate File feature. It takes multiple files and connects them together (the tail of the first file to the head of the next, etc) and defines the resultant file with a unique name. This feature uses the standard controls associated with the Batch File Editor having no ancillary controls of its own.

Batch File Editor (Tutorial)

1. Under the "Filter" menu, click on the "Batch File Editor."
2. Choose a filter that you wish to run a batch of .wav files on by using the left mouse button. *
3. Select the desired filter preset for your batch processing. If the desired settings have not already been saved under a preset

name, you can double click on the filter icon and then set the parameters you desire for the batch processing run.

4. Click on "Add Files."
5. Select the desired .wav files from the file manager. The file listing will appear in the .wav file-listing box.
6. To delete a particular .wav file from your batch-mode listing, click on it, which will highlight it, and then click on "Delete Files."
7. When your list is complete, click on the "Run Batch" button.
8. The Batch listing can be saved by clicking on the "Save Batch" button.
9. To recall a particular batch, click on the "Load Batch" button and select the desired batch using the file manager.
10. If you have a lot of files to process and you are using Windows 98 SE or Windows ME, you may want to check the box labeled "Close Files After Filtering". By default, the files are left open in the editor window after the batch file is run. This can cause Windows to run out of resources when many files (>20) are processed. Checking this box will close each file after it has been processed which removes the resource limit.

Note: If multiple filters are to be run, you can use the Multi-Filter to create your filter sequence.

EZ Clean Filter®



EZ Clean One Step Scratch, Crackle, Hiss and Hum Removal

Sometimes it is very desirable to be able to clean up a piece of audio quickly without any hassles. The EZ Clean Filter® is designed to do just that. It removes scratches, clicks, crackle, static, hiss and hum with one very simple user interface. It includes four time domain visual Noisegraphs® to help tune it up, coupled with only three controls and a few checkboxes making it extremely easy to use. The tradeoff associated with the EZ Clean Filter® is that the clean-up process may not be as finely optimized as can be accomplished using the stand alone filters, each having a full array of controls over numerous variables. But, for a lot of routine audio clean-up work, you will find that the EZ Clean Filter® to be very easy to use.

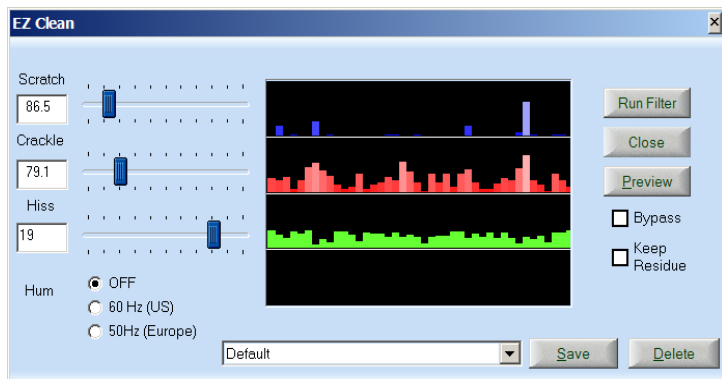


Figure 50 - The EZ Clean Operating Table

Scratch Control

The top control on the EZ Clean Filter® is used for attenuating large clicks, pops, and scratches in the audio material. The normalized range for this control is 0 to 100. A good starting point for this control is 50. To “tweak” (or fine tune) this control (or any of the EZ Clean Filter® controls), use the left and/or right arrow keys of your keyboard. The top graph on the EZ Clean Filter® Noisegraph® indicates the number of events being repaired per unit time vs. time. Taller bar graphs indicate more aggressive filtering.

Crackle Control

This control sets the threshold for the detection and interpolation of small clicks, crackle, static and impulses. Like the Scratch control, low numbers represent its least aggressive behavior, while large numbers provide a more aggressive response. The normalized range for this control is 0 to 100. A good starting point for this control is 50. The Noisegraph® graph for this control is immediately adjacent to its control. Its Noisegraph® also indicated the number of events being repaired per unit time vs. time. Taller bar graphs on the Noisegraph® indicate more aggressive filtering.

Hiss Control

The hiss control attenuates exactly what the name implies as well some continuous noise content over the entire audio spectrum. This system is relatively automatic in that it determines its noise fingerprint “on the fly” and is dynamic. The Hiss control setting will be very much dependent on the audio presented to it, so there is no universal position from which to start. Very noisy material may require relatively low value settings to be effective while relatively quiet material may need a much more aggressive setting. Like the other EZ Clean Filter® controls, its range runs from 0 to 100. Its Noisegraph® graph is adjacent to the Hiss control indicating the aggressiveness of the filter on the material vs. time. If the Hiss Noisegraph® starts to “clip” (hitting full scale), this is an indication that the filter is set too aggressively and will produce “artifacts” in the resultant sound.

Hum Selector

The bottom portion of the Noisegraph® indicates the presence of Hum. This Noisegraph® is only active when the filter has been activated. Red indicates the presence of Hum in the 50 Hz ranges +/- 2 Hz and Orange indicates 60 Hz +/- 2 Hz depending on which mode is selected. Taller bar graphs indicate louder Hum levels, which are plotted vs. time on the Noisegraph®. You must determine the appropriate checkbox to attenuate the Hum, selecting between either 50 or 60 Hz depending on the indication on the Noisegraph® after you do so. (In general, recordings made in Europe will contain 50 Hz Hum while recordings made in North America will contain 60 Hz Hum.) If clicking on one of the two frequencies causes the Noisegraph® to produce a steady stream of Noisegraph® signals, then there is probably Hum present at that frequency. The Hum Filter will remove the fundamental component of the selected value and two harmonics thereof. So if 50 Hz is selected, 50, 100 and 150 Hz components will be attenuated. Similarly if 60 Hz is selected, the system will attenuate the 60, 120 and 180 Hz Hum components.

EZ Clean Ancillary Controls

The EZ Clean Filter® also includes the following ancillary controls:

- **Run Filter:** Clicking on this button will run the filter on the selected file.
- **Close:** Clicking on this will close the EZ Clean Filter®.
- **Preview:** This allows you to hear the effect that the EZ Clean Filter® settings has on the sound as applied to your wavefile. You can adjust the controls while you are previewing in order to achieve the optimum results later when you decide to “Run” the filter.
- **Bypass:** Clicking on this bypasses the EZ Clean Filter®, allowing you to hear the file both before and after (A to B audio comparison) filtering.
- **Keep Residue:** This allows you to hear the noise components being removed by the filter.
- **Presets:** A number of descriptive factory presets are provided which you can use as a getting started aid when using the EZ Clean Filter®. You can also create and save your own presets.

Multi-Filter



DC8/DC FORENSICS provides the capability of cascading up to 20 filters together and running them as if they were a single filter. The elements of the string of cascaded filters can all be unique, or repetitions of the same filter or a combination thereof. This allows you the flexibility to construct your own favorite sequence of filters, along with their presets, and be able to save the cascade along with all of the filter parameters under one user defined single preset name.

The filter combination thereby created can be previewed or run like any other single filter. In Live Preview mode (real time feed-through) you can also run all of the filters in the chain through a full duplex sound card, using the computer as a digital signal processor. For details on this mode of operation, please refer to the section on “Live” mode below.

The best way to envision the Multi-Filter is to merely view it as another DC8/DC FORENSICS filter that can be customized and sequenced. To create a customized sequence of filters merely click on the Multi-Filter icon or select Filter/Multi-Filter, and the Multi-Filter dialog box will pop up. On the left, you will see the source input to the system. On the

right of the screen, you will see the output. The input is that .wav file which is present in the source window, (or in the case of Live mode, it can be the sound card input signal) as is the case with any other filter. The output of the filter can be previewed bypassed, or run just like any other filter. It can be run in Live Preview mode as well (see below).

To create your filter sequence, merely drag and drop the desired filters from the filter selection into the signal pathway in your prescribed order. If you want to change the order of the filters in the cascaded chain, merely click on the filter in the pathway, and drag it to a new location. To get rid of a filter, simply drag it away from the signal pathway with the left mouse button releasing the button when the filter is away from the signal stream. To view the parameters of any given filters in the Multi-Filter, double click on the filter icon in the cascaded chain and the particular filter dialog box will pop-up. Adjust the parameters to the values that you desire, and they shall be saved as part of the sequence of filters that you have just constructed. As with all of the filters, the parameters may be previewed while adjusting the parameters. Be aware that when long chains of filters have been created, there will be a delay in time before the controls react to your changes. If you desire to delete a filter, drag the filter out of the chain and release the mouse button.

After you have developed a chain that you find to be particularly effective for a certain type of task, you can save its settings under a descriptive name just like any other DC8/DC FORENSICS filter for ease of recall later.

It is important to note that improved audio performance will be achieved using the Multi-Filter as opposed to a sequence of .wav file processing. The reason is that there will not be the quantization error buildup since the Multi-Filter will maintain the highest possible resolution for the signal throughout the process from the first filter to the last one in the sequence. Sequential step filtering must always convert back to a .wav file as the intermediate step between filter operations, producing a buildup of quantization errors.

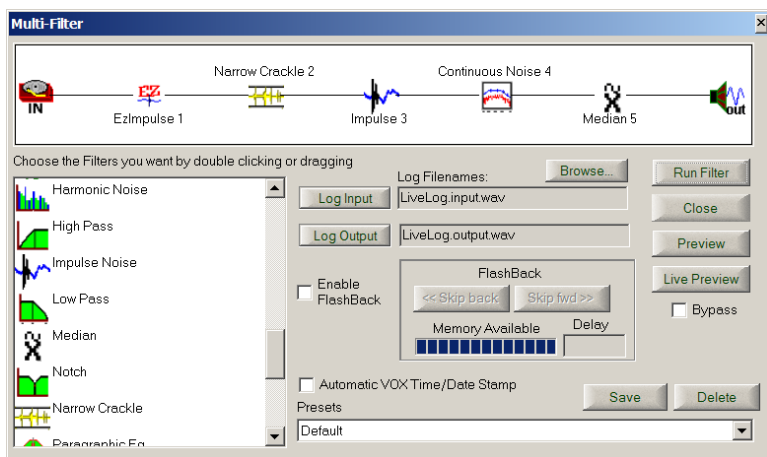


Figure 51 - Run Multiple Filters Simultaneously!

Live Preview

Live (feed-through) mode allows you to bypass the hard disk recording process, making your computer into a real-time feed-through digital signal processor. Signals are fed into your sound card, processed through the filter or filters chosen, and then fed back out of the output of the sound card for use "live." Live mode is useful for professional applications such as broadcasting, or surveillance work where a signal needs to be processed in real-time without the intervening process of hard drive recording. It is also very handy for use in a Ham Radio, Short wave Listening station or even as a permanent part of a home audio system. This feature is particularly useful for real time audio restoration as may be required by radio stations desiring to play old vinyl or 78s directly on the air. It is also useful for cleaning up live news audio feeds or bad audio on talk radio shows.

This feature requires a full-duplex sound card and a fast computer. We recommend the fastest computer that you can afford, since the faster that your computer can perform the mathematical functions; the more filters that you can cascade in "Live" mode. System "stuttering" is an indication that you have exceeded your computer's ability to keep up with the data processing in real-time. Also, latency (the time delay for

processing) is reduced as the speed of your computer is increased. For professional applications, we recommend a 450 MHz Pentium 2 or higher for "Live" mode when used in conjunction with the Multi-Filter feature, although slower computers will run with increased latency and a smaller number of cascaded filters.

Live mode is accessed by clicking the button in the Multi-Filter window. To adjust the input parameters for your sound card, double click on the input icon (the turntable) and a dialog box will pop-up. The following input parameters are adjustable:

- Mono or Stereo
- Sampling Rate (up to 192 kHz)
- Resolution (8 bit to 24 bit)
- Input level indicators are also provided

Adjust the parameters appropriately. Keep in mind that stereo signals will consume about twice the computer resources compared to monophonic signals. So, if you do not need stereo, do not use the Live mode in stereo, else you will be adding extra latency to the signal and reduce the number of filters which can be run without stuttering.

Drag and drop the filters of interest into the signal path. Adjust their parameters by double clicking on each filter, producing its dialog box.

To run the "Live" mode, merely click on the "Live Preview" button, and the signal will be processed through the filters in the signal path. To adjust the output level, double click on the output device (the loudspeaker icon) and a dialog box will appear to facilitate this function. You can also make .wav files in real time using the Live mode. This feature is also contained in the output device dialog box.

Important Note: When using "Live" mode, it is not advisable to connect the soundcard I/O in an effects loop on a mixing console. Doing so can produce echo effects due to the latency of the computational process. Instead, connect the soundcard directly in the signal pathway, which you desire to process.

Live Log to Disk Mode (DC Forensics Version Only)

You can now "Log to Disk" (to your hard drive) both the input and/or output content of the signal path being used in Live mode via the Multi-Filter. These features can be useful when it is necessary to create an archival recording of a surveillance or a broadcast session. It is possible to log both the unprocessed and processed data simultaneously to your hard drive for later retrieval. To perform the Log to Disk functions, merely click on the appropriate Log button(s) that you set up. To log the raw source data, click on the "Log Input" button. To log the processed data, click on the "Log Output" button. You can log the Input and Output signals simultaneously if you desire by clicking and therefore activating both logging functions. Whenever you click on the "Live" button, whatever signals are being presented to the Multi-Filter will be logged to the disk under the Filename(s) indicated next to the "Log Input" and "Log Output" toggle buttons.

Both input and output logging can be stopped and started during a live session, and neither has to be enabled when live mode is first started. Log to disk data will be appended to the files that are shown in the log filename boxes. If the Multifilter is closed and re-opened, a new temp filename will be selected in the log filename boxes. The Automatic VOX and Time/Date Stamp feature applies to both the input and output logging function when checked.

Live Mode VOX Time/Date Stamp (Forensics Version Only)

By checking the box labeled Automatic VOX Time/date Stamp, DC Forensics 8 will add a marker each time recording starts in VOX mode. This is useful in remote location or surveillance recording. By enabling this function, the system will record only when audio is actually present to be recorded and each recording event will be marked with the exact time and date of the individual recording. This function is useful only in Live mode.

Flashback (DC Forensics Version Only)

Flashback lets you quickly Review and Clean an audio stream Live without interrupting your recording

Flashback is primarily designed to be used in real time intelligence gathering and Forensics surveillance situations. It is a feature that allows you to go back and listen to something that you may have heard during real time operation of the software that may be of immediate interest and needs rapid clarification. This feature is available when you are operating the Multi-Filter in Live mode and is only available in the Forensics version of the software. It uses RAM to store up to 300 seconds (5 minutes) of audio data in stereophonic 16-bit, 44.1 kHz format on a FIFO basis. Therefore, it will require about 53 Mbytes of available RAM in order to properly function. It can be operated with or without the log to disc function(s) activated. Faster sampling rates used in the Multi-Filter recorder will de-rate the available Flashback time availability. It always operates on the output signal of the Multi-Filter. Therefore, signal processed data as set up in the Multi-Filter is fed into the Flashback buffer memory.

The controls for the Flashback feature are as follows:

- **Enable Flashback:** Check this box to Enable the Flashback Function
- **Skip Backward:** Moves the play pointer backwards in 5-second increments per click of the mouse.
- **Skip Forward:** Moves the play pointer forwards in 5-second increments per click of the mouse. (Of course, this function only is operational after you have Skipped Backward first, else you would have a time machine on your hands)
- **Delay: (Seconds)** This displays the amount of time shift backwards from real time that you are monitoring via the Flashback memory.
- **Space Available:** This indicates the relative amount of RAM which is available for the Flashback function which depends on the amount of “Skip Backwards” delay you have set.

Available Flashback Time as a Function of Sampling Rate				
Sampling Rate	44.1 kHz	48 kHz	96 kHz	192 kHz
Available Flashback Time	5 Minutes	4.5 Minutes	2.3 Minutes	1.2 Minutes

DirectX Filters

DC8/DC FORENSICS includes the ability to accept standard DirectX plug-ins. There are thousands of compatible plug-ins available which offer many types of audio processing routines. Plug-ins range in price from free to super expensive and are available over the Internet.

To get started, just choose the Filter Menu and then the DirectX Filter. A screen similar to the one below will then appear. Like all our tools, you have the standard “Run Filter”, “Preview”, “Close” and “Preset” buttons available.

Before you attempt to apply a DirectX tool, you must add it to the list of available routines. This list will be presented in the white area on the left side of the display. To add a DirectX tool, just click the + sign and your software will automatically find all compatible tools on your hard drive and make them available. You can add as many as you want from this list.

The – sign removes tools from the “Availability” list and the arrow symbol moves them up and down.

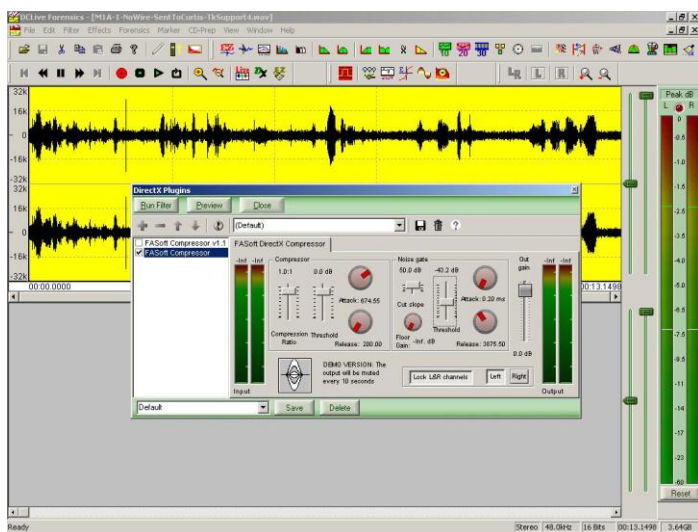


Figure 52 - The DirectX Window

Once you have added tools on the leftmost white area of the window; you can enable them for actual use by clicking in the desired plug-ins checkbox. Only tools that are checked will be applied or previewed. Please note that you can apply multiple tools by checking more than one checkbox.

Clicking on a tool will bring up that specific plug-in's graphic controls in the rightmost areas of the screen. Since DirectX tools are written by programmers other than the authors of DC Forensics, we can't predict what you will see on this screen.

Usage from this point forward is like any other built in tool with DC DC Forensics; just preview and adjust or run the filter.

Using DirectX Plug-ins in the Multi-Filter

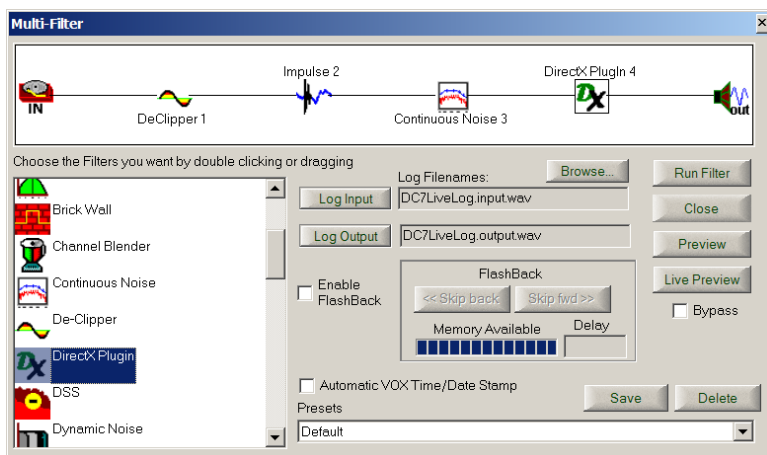


Figure 53 - DirectX Plugs Integrate into the Multi-Filter

As can be seen above, the new DirectX icon is also available in the Multi-Filter so that these types of tools can be freely used with all the native DC Forensics tools. Please see the section in the users' manual describing the Multi-Filter topic for specifics on the use of these tools when cascaded with others.

Caution:

Being able to use audio tools designed by dozens of different companies is a powerful feature in your Diamond Cut software. However, it must be remembered that these tools are all independently designed and created by their respective manufacturers and are not created or supported by Diamond Cut Productions, Inc. Correctly written DirectX Plug-ins should work fine in the program, but it is possible to download a poorly written or buggy plug-in. If you find that a specific plug-in is behaving strangely, we recommend simply deleting or uninstalling it from your hard drive.

Impulse Noise Filters (Clicks, Ticks, Crackle, Snaps Pops and Thuds)

Four Impulse Noise filters are provided in the DC8/DC FORENSICS suite. The Expert filter provides a very high degree of adjustability for those who want to customize the system to their own personal preferences. The EZ Impulse Noise Filter provides automation to simplify the process, with the tradeoff of reduced user controllability. The Narrow Crackle filter specializes in handling impulse noise having narrow pulse width. The Big Click Filter specializes in very large clicks and thuds only. All of these filters are non-linear algorithms used to eliminate pops, ticks, clicks, and crackles from audio recordings. It is also useful for the elimination of "static" interference from AM, FM, or Short Wave radio broadcasts. All of these types of noise signals generally look like impulses (although sometimes referred to as "spikes"), and therefore the name Impulse Noise Filter. The algorithm essentially monitors for fast events, and when their value exceeds a threshold value, the algorithm blanks out the portion of the file wherein the fast event occurred, and re-inserts a waveform that is an approximation of the signal that likely would have occurred during the event. The phase of the inserted signal is aligned to match the point in time in the signal where it is inserted so that there is no phase discontinuity, and therefore almost no artifact will be injected into the .wav file.

EZ ImpulseNoise Filter™



The EZ Impulse Noise Filter combines the noise rejection characteristics of the more controlled performance of the "Expert-Impulse" Filter with some automation and the resultant ease of use. It has only three controls that need to be set by the user. One control determines the aggressiveness of the EZ-Impulse Filter's response to large scratches or impulses. A second control affects its response to "crackle" which is another term for smaller and densely populated impulses. The "Narrow Crackle" control affects its response to smaller impulsive events. Two bar graphs provide the user with visual feedback as to the effect that each of the two controls has on the signal. A tally of "Total Clicks Fixed" is also provided for use as a relative

reference and an operational aid. The following is a listing of the EZ-Impulse Filters controls:

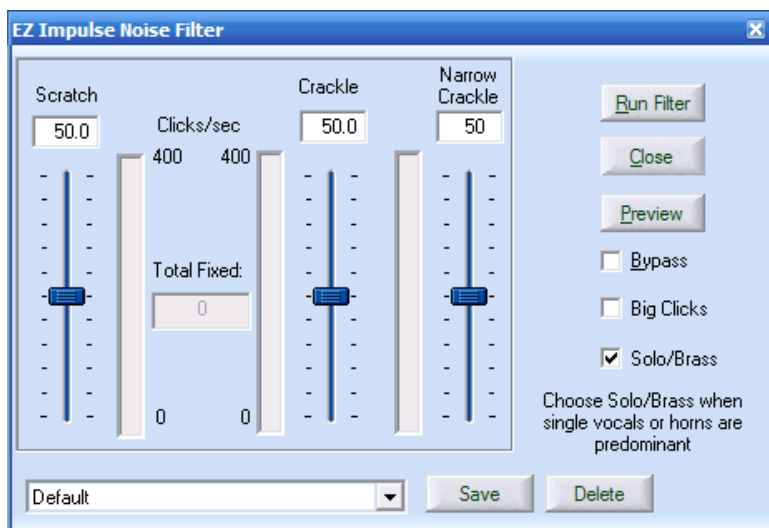


Figure 54 - EZ Impulse- Powerful and Easy To Use!

- **Scratch**

This control sets the threshold for the detection and interpolation of large clicks, pops and scratches. Low numbers are less aggressive, while large numbers are more aggressive. The normalized range for this control is 0 to 100.00

- **Crackle**

This control sets the threshold for the detection and interpolation of small clicks, crackle and impulses. Like the Scratch control, low numbers are least aggressive, while large numbers provide a more aggressive response. The normalized range for this control is also 0 to 100.00

- **Narrow Crackle**

This control sets the sensitivity of the system to narrow pulse width crackle. Like the other controls, increased values represent increased

sensitivity to this type of noise. The normalized range for this control is 0 to 100.00 with a good place to start being around 50.

- **Big Clicks Checkbox**

This feature enables a “Big Click Filter” having a fixed ratio setting of 1.4. For details pertaining to the functionality of a “Big Click Filter”, please refer to the section of this User’s Manual having that name.

- **Bar Graphs**

All three filter controls have their own “Clicks per Second” bar graph having a range of display of 0 to 400. These three graphs are useful for setting their respective controls. Indications that are too low may mean that the system is not sensitive enough to pick up the impulses, while indications that are too high may indicate that the Wave file is being damaged by overreaction of the filter. Using this, in conjunction with the “Preview” mode makes the setting of the EZ Impulse Filter very simple and effective.

- **Total Fixed**

The Total Fixed accumulator counts the total number of impulsive events repaired by the system. It responds to the Scratches and Crackles, but not the Narrow Crackle functions.

- **Bypass**

This control allows you to hear the unprocessed signal for comparison purposes.

- **Solo/Brass**

Choose this setting when brass instrument solos or up-front vocal solos are causing distortion when using the EZ Impulse filter. This feature will suppress distortion produced by those instruments or strong lead vocals. It should not be used on material that does not contain up-front brass or up front vocals as the filters performance will be compromised under those circumstances.

EZ Impulse Noise Procedure (Tutorial)

The EZ Impulse Filter operating procedure is as simple as one, two, three.

1. Place both the “Scratch” and “Crackle” and “Narrow Crackle” controls to their lowest positions.
2. Using “Preview” mode, adjust the “Scratch” Control upwards for the best balance between effective scratch removal and minimized distortion. Generally, that will occur at a value around 55.
3. Next, advance the “Crackle” control upwards for the best balance between effective crackle removal and minimized distortion. Generally, that will occur at a value around 50.
4. Lastly, if there are any residual tiny crackles left behind, raise the Narrow Crackle control until they are attenuated by the system. Generally, that will occur at some value between 20 and 80.

Also, an array of presets is provided with the EZ Impulse Filter that may produce reasonable results without the need for adjusting the filters controls. However, the optimum results will usually be obtained by following the above-mentioned four-step procedure since impulse noise removal is source material dependent.

Note 1: Use the Impulse Noise filter first in your audio restoration process.

Note2: Improved performance of this filter can be achieved by recording at a 96 kHz sampling rate. You can down-convert to 44.1 kHz immediately after the de-clicking process if you so choose, since the remaining filters will not specifically benefit from a higher sampling rate.

Note 3: Up-Sampling a file to 96 kHz that was originally sampled at 44.1 KHz or lower will not provide the same benefits outlined in Note 2. The EZ Impulse filter primarily benefits from a 96 kHz sampled file that was originally digitized (transferred) at 96 kHz from its analog source during the Analog to Digital conversion process.

Expert Impulse Noise



As your expertise with the product grows, so too will your demands from the filters. The Expert Impulse Noise Filter approach allows you to control all of the parameters associated with a basic impulse filter and therefore obtain the most flexible results. It does require a larger degree of experience to use compared to the EZ Impulse Filter. Generally speaking, the Expert Impulse Noise Filter is designed to remove impulsive noises from a signal such as clicks, ticks, scratches, pops and static. It can also be used to attenuate harmonically rich “buzz” from a recording. The Expert Impulse Noise filter is primarily intended to be used in extremely unusual phonographic situations or for Forensics applications in order to filter out buzz or static from a communications link. You will find that the EZ Impulse filter solves almost all impulsive noise problems that you will encounter, but for the rare few that occasionally pop up. So, try the EZ Impulse filter first before experimenting around with the Expert Impulse Filter. The following is a summary of the Expert Impulse Noise Filter control parameter functionality and their adjustment ranges:

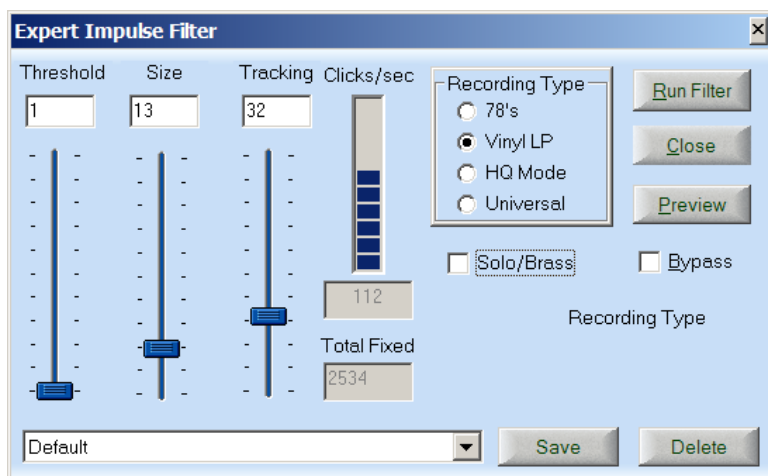


Figure 55 - Expert Impulse Noise Filter

- **Threshold**

This is the Voltage derivative signal level above which the program decides that an impulse noise event has occurred. It has a range of adjustment from 1 to 18,000 (in DAC counts normalized to a 16 bit system). A good starting point for the threshold value is 1/3rd of the full-scale value of the envelope of your .wav file. For example, if the .wav file is almost full scale in amplitude (+/- 32,000 counts) set the threshold at around 10,000.

For 78 RPM records, (with the Tracking control set to 0), start with a Threshold value of 1000, and adjust up or down depending on the results obtained. Lower values of Threshold will produce a higher degree of de-clicking. If it is set too low, however, you will produce distortion on your recording. If you do not want to start at 1000, use the 1/3rd of full-scale rule for your initial setting.

Note: Threshold should be set to its lowest value (slider down) for Vinyl LP/45 RPM applications. Adjustments should be made using the Tracking adjustment for these applications.

- **Size**

This is the number of samples during which the "click" or "pop" event must remain to be defined as an impulse noise event. It has a range of adjustment from 2 to 60 samples. Short "clicks" require a smaller setting compared to longer "pops." A good range of values to start with for Vinyl LP applications is in the 10 to 15 area. A good range of values to use for non-Vinyl (like 78-RPM records) is between 3 to 7 samples.

- **Tracking**

This is the value of rectified output Voltage from a High-pass filter that is used to modulate the threshold Voltage of the filter. When there is a lot of high frequency information present on the recording, like the crashing of cymbals, it is desirable to move the threshold higher in value so that the transients contained in such sounds are not mistaken to be impulse noise events. Tracking has a range of adjustment from 1 to 100 (in relative units). Tracking is most useful on "high fidelity" recordings that contain a lot of "real" high frequency information such as loud dynamic cymbal crashes, or exaggerated sibilant sounds, which may be interpreted by the Impulse Filter as impulse transients.

Most 78s de-click best with the tracking turned all the way down (to a setting of 1). For LPs, start with a setting of around 25 to 30, and adjust the value upwards if distortion is heard on high frequency passages or sibilant sounds, until the distortion disappears. However, if the tracking is set too high, adequate de-clicking may not be obtained.

Note: Tracking should be set to its lowest value (slider down) for non-Vinyl LP applications. Adjustments should be made using the Threshold adjustment in these applications. If the threshold control is set too high, the Impulse filter will not completely de-click your recording. If it is set too low, the filter will create distortion on your recording when high frequency signals are present, especially on the higher registers of the audio scale.

• **Preview Mode**

When enabled, this allows you to hear quickly the results of your chosen settings. If your computer is a slower model, the system may "stutter" when this is enabled, however, the feature can still be quite useful for finding the best settings, since the "stutter" will not appear on the final product. The slider controls can be adjusted "live" when the preview mode is enabled. Preview mode is invoked merely by clicking on the "Preview" button on the filter dialog box.

• **Vinyl LP Mode**

When this is enabled, a different type of detector algorithm is utilized which is more optimized for the wider bandwidth and narrower clicks and pops encountered with Vinyl LPs. Vinyl mode works most effectively on .wav files that are sampled at 44.1 kHz. When this mode is turned off, the detector is more optimized for slower and wider impulse noise. Unlike most of the other controls, you cannot switch Vinyl LP mode on or off "live" when preview mode is invoked. Use the following table for determining the correct mode for this selector based on the type of material you are working with:

Sound Source	Vinyl LP Mode
Vinyl LP (Stereo or Mono)	"On"
45 RPM	"On"
FM Impulse Noise	"On"
FM Stereo Impulse Noise	"On"
Acoustical 78s	"Off"

Electrical 78s	"Off"
Cylinders	"Off"
Hill and Dales	"Off"
Movie Soundtrack "Pops"	"Off"
AM or Short-Wave Static	"Off"
Forensic Audio	"Off"

Note: When you run the Expert Impulse Filter, a dialog box will appear which indicates the Clicks/Second and the Total Clicks Processed. The Clicks/Second statistic is relative to the timing of your .wav file, and not the time frame in which your computer is processing the data. This feature is provided to help you determine if you are "trashing" (creating distortion) your .wav file due to Thresholds or Tracking values that are set too low. When the algorithm's parameters are set too aggressively, the Clicks / Second number will become extremely high, which could be an indication of impending distortion of your .wav file (although it depends largely on the condition of your Source material).

- **Hind Quaternion Mode (HQ Mode)**

DC8 and DC FORENSICS have included an enhancement to the Impulse filter called Hind Quaternion mode (a name related to the method used in the click detector portion of the Impulse filter). This mode can be used with 78s, vinyl, or any source containing impulsive type noise or clipping distortion. Its operation couples several related variables in this new detector algorithm together in a logical ratio-metric relationship allowing you to vary them with one control. The Size slider controls these new variables. The other controls on the Impulse filter still perform their previously defined functions. In HQ mode, you will find that you have a much larger degree of control of the detector algorithm, especially through the use of the Size control. Small fast rise time clicks will be detected optimally with small values of Size while larger and slower events will be best detected with larger settings. The advantage of this new mode is an improvement in the ability of the detector to reject musical transients while still maintaining good click sensitivity.

- **Universal Mode**

Universal Mode provides an adaptive element to the detector portion of the Expert Impulse Filter algorithm. It can be used with any type of

impulsive noise, whether the source is from records (78s, 45s, 33.3s), radio static, or tape head static discharge. Universal Mode is of particular advantage in some wireless surveillance applications where occasional static needs to be mitigated. The advantage of this mode is its ability to adapt to changing signal environments. The Tracking control is its primary means of adjustment. The Tracking control affects the systems overall centered value of sensitivity while the Threshold control sets the minimum value that the system can apply to the detector. The size control does the same thing as with the all of the other modes associated with the Expert Impulse Filter. The disadvantage of the Universal Mode detector is its diminished transient response. This will cause the system to have some difficulty in dealing with the leading edges of rapidly changing audio material, since it will take some time to re-adjust itself. This could cause some transient leading edge distortion and/or leading edge missed impulse detection.

Important Note: The different Impulse Noise filter modes (LP, 78, HQ, Universal) all take a somewhat different approach toward identifying clicks and pops. Since these types of impulses can come in an almost infinite number of sizes and types, a user may select the mode that provides good results over a very wide range of impulsive noise environments.

- **Solo/Brass Mode**

DC8/DC FORENSICS provides you with a discriminator routine which identifies brass instruments and excludes them as candidates for interpolation. Since brass musical instruments (like solo trumpets) produce waveforms that are quite similar in form to impulsive noise events, this option can be quite useful in preventing distortion from being interjected upon those types of instruments. This mode of operation is also useful on some “close-miked” vocals where the background instrumentation is low in sound level comparatively speaking. The tradeoff with using Solo/Brass mode of operation is reduced de-clicking capability during the actual presence of the brass instrument or vocal in question. This mode should not be used on material that does not contain strong solo brass instruments or up-front lead vocals.

Note 1: Do not mute the beginning or the ending of a .wav file before operating the impulse noise filter. Mute the extraneous noises from the

beginning and the ending of your .wav file at the end of all of your audio restoration processes.

Note 2: There will be occasions wherein 78-RPM recordings will benefit from the de-clicking action of the vinyl mode impulse filter in conjunction with the tracking control. If the clicks are small and short, it is worth giving it a try.

Note 3: Multiple passes through the impulse filter, especially in vinyl mode while using the tracking control, can produce ever-improving results.

Expert-Impulse Noise Operating Procedure (Tutorial)

1. Highlight the portion of your .wav file on which you desire to apply the Impulse Noise Filter. (You may choose to highlight the entire file or any portion thereof.) Sometimes, when confronted with extremely stubborn clicks or pops, or radio "static" it may be useful to use the Zoom-In feature first before running the Impulse Noise Filter on a "grouping."
2. Click on the "Filter Menu" with the left mouse button.
3. Click on "Impulse Noise"
4. Start with the "Threshold" control at a setting of approximately 1000 for 78s.
5. Start with a "Size" setting of between 3 to 7 samples for non-Vinyl applications, and use a setting somewhere in the 10 to 15 sample range for Vinyl LP and 45 RPM record applications.
6. If you are de-clicking a Vinyl LP record, click "Vinyl LP" on with the left mouse button.

Note: This feature is also utilized for 45-RPM records. If you are de-clicking a 78-RPM record or something similar, make sure "Vinyl LP" is turned-off. (It is important to note that Vinyl LP mode works best on .wav files that have been sampled at 44.1 kHz or higher.)

7. If you are de-clicking a Vinyl LP record, set the threshold control to its lowest value, and perform all of your adjustments with the tracking control, starting with a setting of 25 to 30. If you are de-clicking a 78-RPM record or something similar set

the tracking control to its lowest value and perform all of your adjustments with the threshold control.

8. Click on "Preview."
9. Listen to the "Previewed" version of the processing parameters that you have just set. If your computer is too slow, it will "hick-up" or "stutter." (Do not be concerned that your final sound restoration will sound like this, since it will not!) Try to listen "through" the stutter to judge what the Filter is doing. If the "stutter" is too annoying to make a judgment of the performance of the filter settings use Run filter mode on a selected portion of the .wav file directly into the "Destination" workspace. Alternately, run the filter, and then listen to the Destination Workspace in order to make judgments regarding your settings. Iterate until you are satisfied with the results.

Note: Setting the filter too aggressively may cause excessive stuttering during Preview. Even a fast computer will stutter when this filter is finding hundreds of clicks per second. If this is happening, readjust the filter to be less aggressive.

10. When the filter is running, you will see a display of "Clicks / Second" and "Total Clicks Processed." Generally speaking, when the threshold is set too low, the program will begin to react to sound transients rather than just noise transients. If the "Clicks / Second" is greater than 30, there is a good chance you are catching sound transients, and creating distortion on the output of the filter. Most records will show less than 10 clicks per second when the settings are correct (except in extreme circumstances). Keep adjusting the threshold setting until the clicks are being removed and distortion is not being produced on the filter output. (The distortion that can occur will be most prevalent on the sibilant sounds.) Keep in mind that lower value settings of the threshold control will cause the algorithm to be more sensitive to removing clicks and pops. However, if it is set too low, distortion will also be produced on the sibilant sounds.
11. If the algorithm is capturing the larger impulses but not the smaller ones, try decreasing the "Size" adjustment, and re-evaluate the results. (You may also have to decrease the threshold control.)

12. When you determine the best setting of the controls for your particular .wav file, click Run filter. When the filter has completed its operation, the results will appear in the "Destination" workspace.

Note: Notes 1, 2 & 3 found at the end of the EZ Impulse Filter Tutorial also apply to the Expert Impulse Noise Filter.

Narrow Crackle Filter



The Diamond Cut Narrow Crackle Filter (NCF) is a special impulse filter designed to identify and interpolate narrow pulse-width crackle type noise. It is especially effective on mint condition LPs having only very small impulsive noise due to vinyl defects or static discharge. This filter is also useful on other recording types provided that larger impulsive noises have been removed by one of the other Diamond Cut Impulse Noise filters first. Additionally, it can be used to remove some power-line frequency related high frequency “buzz” (which is high frequency & harmonically rich “hum”). The advantage of the Narrow Crackle filter is that it is the least invasive of all of the impulse noise reduction filters in the Diamond Cut suite. We have found it to be effective in reducing the residual crackle sounds on not only Vinyl LPs, but also transcription acetate recordings and 78s after all of the other impulsive noises have been removed first. It is very easy to operate, having only two controls which are as follows:

- **Threshold:** This control sets the relative amplitude at which this filter detects narrow crackle impulsive noise having a range of 5 to 100. Adjust this control until downwards until the impulsive noise is reduced but not so low as to reduce the bandwidth or produce distortion on the target signal. A good place to start with on this control is a setting around 30. Settings above 90 can produce distortion and also a Low Pass Filter effect on the signal.
- **Size:** This parameter sets the pulse width to which the system is sensitive having a range running from 1 to 15. Larger values of size represent longer pulse width values. Use the smallest value of size to do the job at hand. A good value to start with is around 3 to 5.

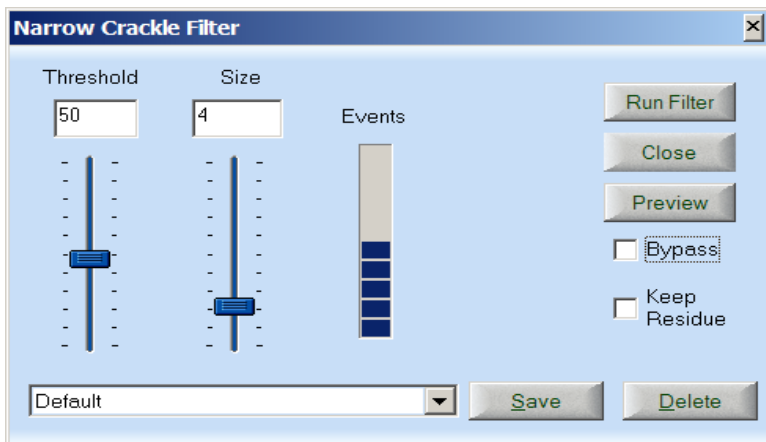


Figure 56 - Narrow Crackle Impulse Noise Filter

The Narrow Crackle Filter’s “Events” bar graph provides you with a relative indication of how aggressively the filter is operating. It “modulates” vertically in proportion to the number of events that are being interpolated by the system. The “Keep Residue” feature allows you to hear what the Narrow Crackle Filter is removing from your source signal. A good way to “tweak” this filter is to highlight a short sector (around 10 seconds) of the .wav file that has the type of narrow crackle impulse noise that you desire to eliminate or attenuate. Preview that sector and adjust the two controls for the best result on that 10 second sector. Satisfied with your settings, it should then do a fine job on the entire file in “Run Filter” mode.

Big Click Filter



The Big Click Filter (BCF) handles exactly what the name implies which includes the elimination of very large clicks but also loud thuds. Its response intentionally ignores smaller clicks and ticks. Its response covers the long time interval range of events lasting greater than 2 mSec but less than 200 mSec. Its detector is actually sensitive to the time amplitude product of the applied signal. This parameter is user adjustable by way of the “ratio” control which has a range which can be varied between 0.75 and 2.5. 1.4 is a good starting place with higher

values decreasing the filters aggressiveness and lower values increasing aggressiveness.

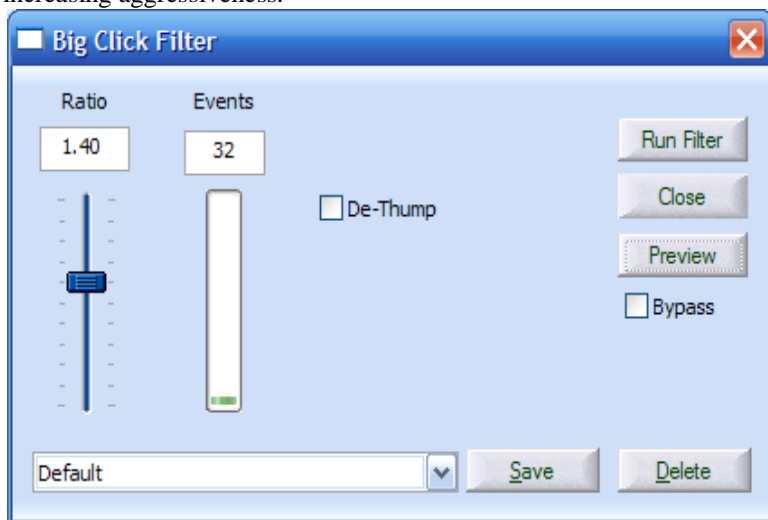


Figure 57 – Big Click Impulse Filter

This filter is very useful for fixing big clicks caused by phonograph records that are cracked or have been glued back together or have major gouges on their surface. It is also very useful for interpolating some of the long lived noises found on 16 inch acetate transcription recordings. Finally, it has a Forensics audio application in that it can attenuate cell phone noise interference from audio recordings. Big clicks should always be removed as the first step of the de-clicking process before attempting to remove smaller impulsive noise using any of the other Impulse filters. The interpolation portion of the algorithm uses a frequency domain technique. Big Click Signals which last up to 200 mSec are able to be handled. The minimum impulse size of 2 mSec is handled by the BCF. Very large clicks and thuds sometimes excite the resonance of a tone arm leaving behind a long low frequency “tail” or thump. This tone arm resonant thump can be dramatically reduced by checking the checkbox labeled “De-Thump” which rejects the “tail” for a period of time of 150 mSec past the end of the interpolation portion of the Big Click Filter algorithm. It is important to note that the “De-Thump” function will only remove “Thumps” that

are preceded by a large click event; it is not capable of removing stand alone thumps from recordings.

The Big Click filter has the following controls and indicators with the following ranges of adjustment:

- Ratio: 0.75 to 2.5 (Lower Values produce more aggressive results)
- Events: Records the number of events found and repaired by the filter
- Events Bar Graph: Indicates the number of events being found and repaired per unit of time.
- De-Thump Checkbox: Attenuates resonant ring-out tails following the repair time interval associated with this filter. Only use this function for the repair of large clicks associated with cracked or broken phonograph records. This function can't be "hot-switched" while previewing.

A demo .wav file is provided in the Wavefiles folder within your Diamond Cut Directory called "BigClickCracked78Demo.wav". Use this file to experiment with the Big Click Filter.

Note: For best results when using the Big Click Filter in the Multifilter, always place it first in line in front of any other impulse filters (or other filters or effects).

Continuous Noise Filter



This filter is useful for reducing background "Hiss" and other constant noise from a recording or from a noisy FM radio transmission. It is referred to as a "Continuous" noise filter (or CNF) because unlike impulse noise, hiss is present at all times. When adjusted properly, this filter can almost completely eliminate all residual noise from a recording. However, it is easy to overuse this filter and leave the recording sounding dead and lifeless, and also introduce digital artifacts into the music.

The Continuous Noise Filter has 4 modes of operation to choose from:

- Normal CNF Mode
- Spectral Subtraction Mode
- Auto Spectrum CNF Mode
- Forensics AFDF Mode

To use this filter in Normal CNF or Spectral Subtraction modes, you must first take a sample of a section of noise. This noise template will then be used by the algorithm to decide what constitutes noise and what constitutes music or speech during the filtering process. It is important to sample a section of the .wav file that does not contain any music/speech so that the filter does not remove signals that contain musical or speech information.

Note: We strongly recommend trying the new Artifact Suppression mode when using this filter; its performance is often much better than standard mode.

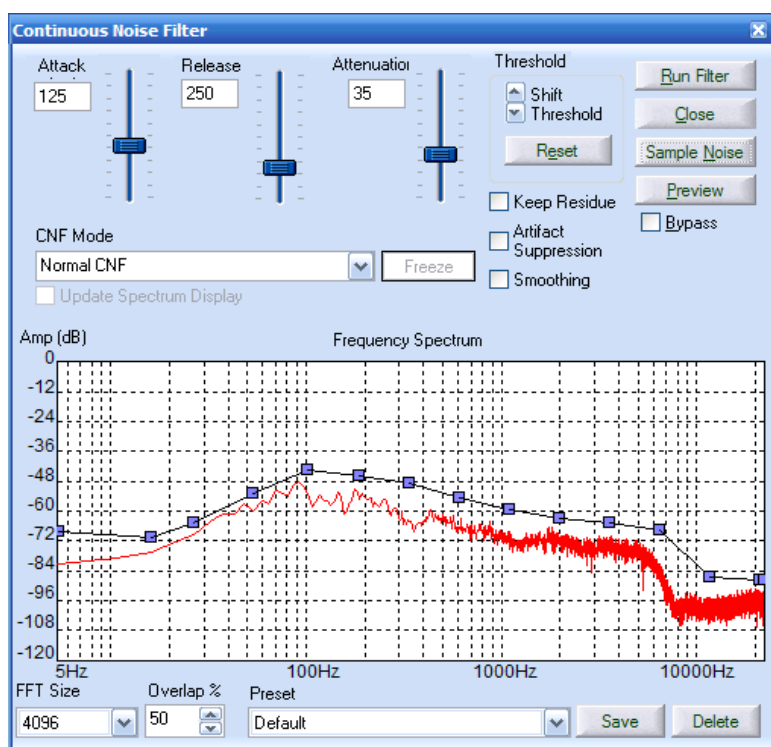


Figure 58 - Continuous Noise Filter

The filter graphically shows a frequency spectrum of the sampled noise in red (sometimes referred to as a noiseprint). This spectrum represents the amount of noise at each frequency band in the recording. The blue line represents the filter design that has been accomplished by the program. You can use the mouse to move the blue threshold line to tailor the kind of noise reduction that the filter performs.

This filter should only be used on recordings that have little or no impulse noise, or on recordings that have already been processed through the Impulse Noise filter in order to minimize the possibility of producing digital artifacts. When operating this filter in the Auto Spectrum CNF Mode, the system will automatically find and modify its own noise fingerprint on-the-fly. Therefore, there is no need to

manually take a noise fingerprint when operating in the Auto Spectrum CNF Mode.

This is one of two different types of non-linear filters that can be used to reduce noise from a signal source. Like the Dynamic Noise Filter, it is useful for reducing "Hiss" from a recording or from a noisy FM radio transmission. However, unlike the Dynamic Noise Filter, it will also reduce lower frequency noise. When adjusted carefully, it can almost completely eliminate all residual noise from a recording. However, when compared to the Dynamic Noise Filter, this filter is a bit trickier to adjust so as to avoid the introduction of digital noise artifacts into the Destination .wav file. It also can have some detrimental effects on the "presence" and the musical transient content of a .wav file when not properly adjusted.

This filter takes a sampling of your file and converts it into the frequency domain utilizing a Fast Fourier transform. Next it marches along to the next time interval and performs another Fourier transform. It keeps repeating this process until the entire .wav file is converted into samples which are no longer representing the time domain, but strictly represented in the frequency domain, with the appropriate Voltage, phase and frequency co-efficients for each window contained in memory. The entire audio spectrum is divided into up to 8192 bands by a 16,384 point fast Fourier transform (FFT) algorithm (the resolution of which is user selectable). When a signal in a particular band exceeds a threshold (when operating in the Normal CNF or the Auto Spectrum CNF Modes) that you can define graphically, then that particular band is allowed to pass its signal from the input of the algorithm to the output of the algorithm. Lastly, the entire file is then re-converted back into the time domain via an inverse fast Fourier Transform. So effectively the only time during which bandwidth is provided at any of the selected number of frequency buckets is when there is a useful signal present in any particular bucket. Otherwise, the various frequency bands are substantially attenuated.

Insert Control Points Into the Continuous Noise Filter

By simply right clicking or double clicking anywhere in the Frequency Spectrum of the Continuous Noise Filter, you can now insert your own Control Point. This can help increase the flexibility and power of the

filter. On right click, the menu that appears allows you to add a point, delete a point, or reset all points to -100 dB.

The following is a summary of the control parameters functionality and range of adjustment provided for the Continuous Noise Filter:

- **Attack Time**

This is the time required for any of the filters to "open up" on the leading edge of a signal that exceeds the threshold line on the spectral graph.

This represents the time constant normalized value at 1 kHz. The time constant for filter frequencies operating above 1 kHz will be shorter than the setting, and the time constant for filter frequencies operating below 1 kHz will be longer.

(The Attack time constant value is weighted with a -1 slope across the audio spectrum.) Small values of attack provide excellent transient response, while long values provide a minimization of digital artifacts produced by the system. A good value to start with for the Attack Time parameter is around 125 Milliseconds. The total range of adjustment for Attack time is 10 to 300 milliseconds. Smaller settings will improve transient response but allow more digital artifact to pass through. Larger values will decrease transient response, but will minimize the production of artifacts during the noise reduction process.

- **Release Time**

This is the time allowed for any of the filters to "close down" or "decay" following a signal that falls below the blue threshold line on the spectral graph. All remaining characteristics of the Release time Constant are the same in nature as the Attack time Constant. A good value to start with for the Release Time parameter is around 50 to 100 Milliseconds. If there is too much fast filter "breathing" (a digital artifact that is also sometimes referred to as "pumping"), lengthen this time until you are satisfied with the result. The total range of adjustment for Release time is 10 to 600 milliseconds.

- **Attenuation**

This control sets the degree of attenuation for signals that are present and below the blue threshold line. The greater that one sets the Attenuation control, the greater will be the degree of noise reduction. However, the greater the degree of noise reduction achieved, the greater will be the loss of the sense of "Ambiance" on the resultant recording. So you must make a careful judgment as the correct tradeoff between noise reduction and ambiance for the material you are dealing with. A good value for the Attenuation parameter is around 10 dB to start with. If there is too much loss in signal ambiance, decrease this value. If you desire more noise reduction, increase this value. If you increase the attenuation too much, you will begin to introduce some digital aliasing artifacts into the Destination .wav file. The total range of adjustment for Attenuation is 0 to 100 dB.

- **Threshold (Blue Graphical Threshold Line)**

This feature controls the threshold value above which a signal at a particular frequency must exceed before it is passed through to the output of the algorithm without attenuation. It works in conjunction with the "sample noise" button. Although the continuous noise filter has up to 8192 discrete frequency bands, it would be inconvenient to have to set each of them. DC8/DC FORENSICS provides up to 256 inflection points (shown as blue dots connected by blue lines on the graph of amplitude vs. frequency) that can be moved along both the frequency and the amplitude axis. The software will place these inflection points automatically at approximately 10 dB above the noise floor after you perform the "sample noise" function. Try these settings first to find out if the results are acceptable. Thereafter, if there seems to be some noise that needs more attenuation at a particular frequency, adjust the threshold upwards utilizing the left mouse button at the frequency of interest until you are satisfied with the results. The graphical threshold line is adjustable "live" when preview mode is enabled. If you want more than the 10 inflection points provided by default, double click using the left mouse button while pointing at the desired position on the graph. To remove inflection points, use the right mouse button in a similar manner.

- **Threshold Control Grouping**

- a) Up & Down "Shift Threshold" Control - This feature allows you to globally shift the entire

threshold line up or down independent of frequency. The feature consists of an up and down arrow box. It is operated via the left mouse button. The amplitude resolution of the control is 4 dB / click. After clicking on either of the arrows, you will see the entire threshold line shift in the direction of the chosen arrow.

- b) "Reset" Control - This feature will restore all of the threshold line inflection points to their original default settings.

- **Keep Residue Function**

When enabled, the "Keep Residue" function will allow you to "preview" (hear) or process to the Destination Workspace the algebraic difference between the Source File and the Filters Output. In essence, you will be listening to the noise which would have been removed from the Source File had this function not been enabled. It is sometimes useful via this function to be able to hear just how much of the real audio signal along with the noise components which you are removing from the source signal. However, it is only fair to warn the user not to make final adjustments using this feature, as that technique can be quite deceptive. It is always best to optimize your parametric settings for the best results while listening to the actual processed filter output signal (i.e. "Keep Residue" function off).

- **Artifact Suppression Mode**

This mode of operation reduces the level of digital artifacts (sometimes referred to as "the birdies") produced by the CNF during its noise reduction process, especially when it is being used aggressively. It can also be used to effectively reduce some forms of inter-modulation distortion from a recording (like that "raspy" sound found on some distorted 45 RPM records - - especially on vocals). When operating the CNF in Artifact Suppression Mode, the Attack and Release functions are eliminated in that the routine operates independently of those parameters. The "Attack" control will become grayed out and the Release control will revert to a new mode of operation simply called "Artifacts". The artifact suppression mode is disabled in Auto CNF mode. In artifact suppression mode, use the CNF as you normally would, including the taking of a noise print sample in the other two modes of CNF operation (normal mode and spectral subtraction mode).

The Artifact Suppression control affects the degree to which the system attenuates digital artifacts with higher settings providing a more aggressive action. You should note that higher levels of “Attenuation” are achievable in Artifact Suppression Mode compared to non Artifact Suppression mode. 50 to 60 is a good setting to start with for the Attenuation control with an FFT size set to 4096. Experiment with FFT sizes on each side of the recommended value to find the optimal result on any particular file. 200 is a good setting to start with for the Artifact Suppression Control. Adjust this control upwards for an increased artifact reduction effect and downwards for a reduction in the removal of audio material (ambience and transients, etc). Use this control to find the best balance between those two sonic characteristics of the particular material that you are working with. Note that the optimal settings for the mentioned controls are substantially material dependent. Using the “Keep Residue” mode will allow you to monitor how much audio material is being removed from the signal by the CNF Artifact Suppression system.

The Artifact Suppression system does not work in Auto Spectrum mode but does in Normal Mode and Forensics AFDF mode. Also, it is to be noted that fairly high FFT sizes produce more optimal results with this system. FFT sizes below 1024 are not recommended with the Suppressor because the system is less effective due to the poor frequency resolution associated with reduced FFT's. It is worth noting that Artifact Suppression mode requires much higher levels of CPU resources and thus takes around 4 times longer to process a given file compared to the normal (non-Artifact Suppression) mode.

- **Smoothing Mode**

An alternative to Artifact Suppression mode is Smoothing Mode. The smoothing checkbox applies some additional signal processing that reduces the digital artifacts or “Musical Noise” that may be heard when applying the Continuous Noise Filter by averaging adjacent frequency bins within an FFT. This feature is most useful on very noisy recordings with high levels of surface noise such as old 78s or extremely noisy Forensics situations. When using the smoothing function, you should be able to increase the level of noise reduction without introducing digital artifacts by about 3 dB or more. The tradeoff, and there are always tradeoffs, is that the frequency selectivity

is reduced in this mode (sometimes resulting in reduced bass response). An example of where the smoothing function would not be useful would be when you are trying to remove pure tones or steady state buzzing sounds where you need to maintain a high degree of frequency selectivity. Please note that this function does not apply to spectral subtraction mode and can't be used in conjunction with the Artifact Suppression mode.

- **FFT Size (Resolution)**

Choose between 32, 64, 128, 256, 512, 1,024, 2,048, 4,096, 8,192, and 16,384.

The frequency resolution of the Continuous Noise Filter can be adjusted. This parameter determines the number of frequency bins used by the FFT algorithm. The actual number of frequency bins produced is the FFT Size divided by two (since the FFT produces data for both the real and imaginary planes). Large values of resolution produce the largest degree frequency selectivity and thus high degrees of noise reduction, while the best time domain transient response will be realized with smaller FFT values. Put another way, there is a tradeoff between frequency resolution and time resolution and they are inversely related to one another. You will have to experiment with the various binary weighted values to determine the best resolution for the material that you are dealing with. Listen for the best levels of noise reduction attainable while minimizing any digital artifacts and yet maintaining good musical transient response on things like rim shots on drums, and other percussive instruments.

Important Note:

This parameter cannot be adjusted while the filter is previewing or running.

- **Overlap (Window Overlap)**

This allows you to choose the percentage of overlap between FFT windows. The range for this adjustment is 40% to 50%. Often, reduced values of % overlap will yield better results in terms of digital artifacts produced, but will take longer to process. Conversely, larger values of overlap will produce more digital artifacts, but take less time to process. Start with 50% and lower it order to achieve faster musical

transients tracking. Note that this feature is only available in the DC Forensics Audio Laboratory version of the software. The overlap parameter is fixed at 50% in the standard version of the software.

- **Spectral Subtraction Mode In The Continuous Noise Filter**

This changes the mode of the Continuous Noise Filter into a spectral subtraction system making it useful for some Forensics noise reduction applications. As with the normal mode of the filter, one still takes a fingerprint of the noise when operating in Spectral Subtraction mode. The amplitude signal of the resultant FFT is then subtracted from the entire .wav file signal thereby providing rejection of the highlighted fingerprinted signal. The offending signal could be an air conditioner, city noise, automobile noise, etc. To adjust the depth of effect, use the attenuation control. The attenuation slider sets the maximum amount of attenuation as an absolute number (as in all of the CNF modes) and in Spectral Subtraction mode the threshold shift scales the overall gain of the noise sample. Threshold and attenuation are not equivalent parameters because of the fact that the maximum attenuation value is absolute but it is not a maximum delta value between the signal and the sampled reference.

Important Note:

If the parameters for the Continuous Noise filter are set incorrectly, it has the propensity to produce extremely strange sounds (digital artifacts) - - - some of them quite comical in nature. If you hear these "birdies", back down on the attenuation setting and / or some of the graph inflection points and they should disappear. If not, increase the Attack time and decrease the Overlap percentage.

- **Auto Spectrum CNF Mode**

Sometimes, it may be desirable to have the Continuous Noise reduction system determine its own noise fingerprint "on the fly." There are three possible reasons why you might be interested in using this mode:

- It simplifies the operation of the Continuous Noise Filter
- There may be no discernable quiet passage on the source material from which to "Sample the Noise Fingerprint".

- The Noise Fingerprint may be changing in its noise distribution dramatically throughout the .wav file.

The Auto Spectrum CNF Mode solves these problems by calculating the noise fingerprint “on the fly” and updating it on a continuous basis. It is capable of performing this function even if you can’t see or hear a totally quiet passage in the target .wav file. The mathematical routine utilized to create this fingerprint is more sensitive than your ears and can always find a noise fingerprint. This system may not be quite as effective for removing noise compared to the other two modes of the CNF operation, but its convenience coupled with its ability to adapt to changing noise environments sometimes outweigh its noise reduction performance limitation. To operate this filter, merely preview it and adjust (primarily) the attenuation, attack and release times for the best sounding results. As with all of our filters, the “keep residue” mode can be useful as an adjustment tool. In “keep residue” mode, tune for maximum noise and minimum signal while operating in “Preview” mode. The Auto Spectrum CNF Mode has the following controls and displays:

- Attenuation (start with a setting of 25)
- Attack Time (start with a setting of 20 mSec)
- Release Time (start with a setting of 50 mSec)
- FFT Size (start with a setting of 8,192)
- Smoothing Checkbox (on)
- Overlap % (50 %)
- Dynamic Spectral Display (on/off selector) (your choice)

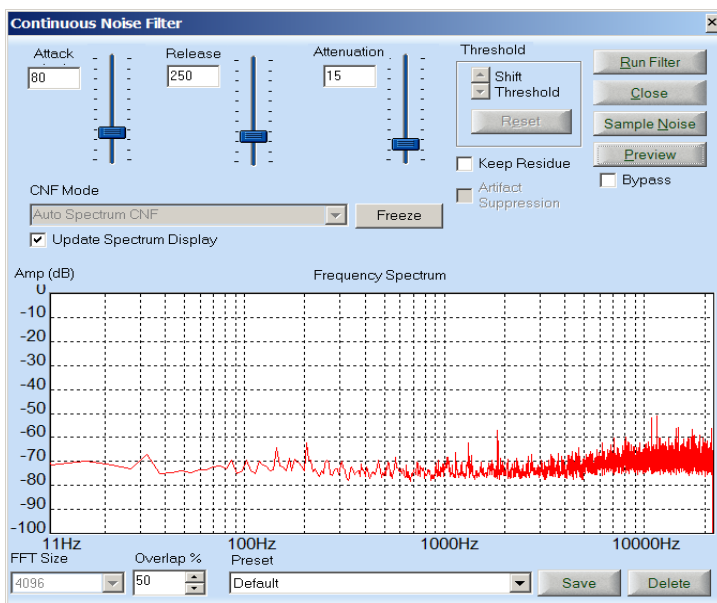


Figure 59 - Auto Spectrum CNF Screen

Forensics Adaptive Frequency Domain Filter (AFDF) Mode (Forensics Version Only)

The Forensics Adaptive Frequency Domain Filter (AFDF) is a variation on the basic theme of the Auto Spectrum CNF filter mode. The primary difference is that the Forensics AFDF is optimized for Forensics oriented files and not “High Fidelity” files. It has a faster response time and a narrower effective bandwidth while producing higher levels of noise reduction at the expense of potentially producing higher levels of digital artifacts. The AFDF Filter is “Adaptive”, which means that it will automatically adjust itself to varying noise environments. This filter has a Time Domain twin sister found in the Forensic Menu, which is simply called the “Adaptive Filter”. The AFDF can produce more attenuation of noise compared to the Time Domain Adaptive Filter but can also produce more spurious digital artifacts. The Time Domain Adaptive filter does not produce spurious digital artifacts, but can produce an echo delay sound as an artifact. We provide both types of Adaptive filters since they exhibit somewhat

differing sonic characteristics. Experimentation is the best way to determine which filter is the most effective on a given file. Also, a combination of the two used together in cascade in the Multi-Filter has been shown to work some magic that otherwise would not be possible on some extremely noisy forensics files. In general operation, the controls and the display graph for the Forensics AFDF is much the same as that of the Auto Spectrum CNF. Refer to it for details.

Continuous Noise Filter Procedure (Tutorial for Normal CNF Mode)

This filter is the most mathematically complex of all of the DC8/DC FORENSICS algorithms. It will, therefore, take the longest amount of processing time to complete its calculations. This algorithm will benefit the most from the use of a high clock rate computer. This filter is also the most difficult filter to use correctly. Aliasing artifacts can be produced when the settings are not correct for the particular .wav file you are attempting to "de-noise." The first time you use it, it will be worthwhile to spend about an hour playing around with it in order to become familiar with its behavior.

1. Highlight a quiet portion of the Source .wav file. Often, this sector will be found at the beginning or at the end of the file, as with the lead-in or the lead-out groove of a record or the lead-in portion of a tape recording. The idea here is to capture a section of noise only, but no signal. This will become the noise floor baseline for the subsequent operation of the Continuous Noise Filter.
2. With the left mouse button, click on "Filter."
3. Next, click on "Continuous Noise."
4. When the Continuous Noise Dialog Box appears, click on "Sample Noise."
5. Some calculations will be made in the ensuing moments. When they are complete, a graph will appear showing the Amplitude (in dB) versus the Frequency of the .wav file noise floor. This graph represents the Noise Print of the file on which you are working.
6. The measured sample noise spectrum is shown in red. The noise threshold value versus frequency is shown in blue. You can set the blue graph threshold value, although DC8/DC FORENSICS will automatically choose some settings for the threshold limit line that is a good place to start.

7. If you choose to change the graphical threshold contour, follow the procedure outlined in steps 7 through 10. Using your mouse, place the pointer on the left-most blue threshold marker on the graph (one of ten blue dots).
8. Depress the left mouse button and move the dot either up or down so that it remains somewhere above the red line graph at the bottom end of the spectrum. The higher this line is from the red line, the greater will be the degree of noise reduction at frequencies near this particular dot. If the dot is placed below the red graphical line, no noise reduction will be applied to these frequencies. This is sometimes the preferable setting for the blue threshold line near the bottom end of the audio spectrum (below a few hundred Hertz).
9. Next, move the next blue threshold marker just to the right of the first one, and using the mouse, set it somewhere above that particular frequency on the spectrum graph.
10. Repeat this process until all ten threshold markers are located somewhere above the "noise floor" graphical representation of your .wav file. Now the blue line should be located above the red line at all frequency locations. Note that the best contour can only be achieved by not only moving the markers along the vertical axis, but along the horizontal (frequency) axis as well.
11. Set the "Attack" time initially to 25 milliseconds.
12. Set the "Release" time initially to 50 or 100 milliseconds. (The "Release" time constant should always be set longer than the "Attack" time constant for a realistic sounding operation of the filter.)
13. Set the "Attenuation" control initially to 10 dB. (Higher numbers results in higher levels of noise reduction.) Too much noise reduction will produce digital artifacts and detract from the "ambiance" of the recording.
14. Highlight the portion of your .wav file on which you desire to apply the Continuous Noise Filter. (You may choose to highlight the entire file or any portion thereof.)
15. Run the Filter.
16. Play the section that you have just processed, and determine which parameters need modification. If there is a "lagging" response to audio signals, decrease the "Attack" time. If there is a "swell" of noise following an audio crescendo, decrease

the "Release" time. If a portion of the audio spectrum is sounding dull, lower the "threshold" line at the frequency of interest. If a portion of the audio spectrum is sounding noisier compared to the rest, then raise the "threshold" at the frequency range of interest.

17. When you are satisfied with the results, re-run the .wav file (in its entirety) to achieve the final processed results in the "Destination" workspace.

Note 1: The threshold line inflection points can be adjusted "live" when you are running the filter in "Preview Mode." You will be able to hear the effects of modification that you make to the threshold line almost immediately.

Note 2: Sometimes, record albums that are in excellent condition exhibit only one type of noise which is called "Rumble". Rumble comes from the record mastering process as well as from the bearings in your turntable. Generally, one uses a High Pass filter to remediate this problem, but that approach also cuts into the low pass portion of the audio signal. There is a special preset in the CNF called "Dynamic Rumble (Only) Filter" which will remove this type of noise without damaging the deep bass on the recording or affecting any other signal above 90 Hz. This filter is more effective on Rumble noise than the standard High Pass Rumble filters found elsewhere in this software program.

Harmonic Reject



The Harmonic Reject filter, which is sometimes referred to as a "comb" or "multiple notch" filter is used to attenuate periodic noises that contain harmonics. It is capable of attenuating either the odd or even harmonics of the selected fundamental frequency. This filter is very effective for attenuating a form of hum (line frequency related) noise that is rich in harmonics. This type of noise is sometimes referred to as "Buzz." Noise, such as this can be introduced into an audio recording from sources such as light dimmers or switch mode computer power supplies. "Buzz" can contain odd harmonics of the power line frequency, all the way through the entire audio spectrum. A 60-Hertz square wave is, by definition, the fundamental component of the

waveform plus all the odd harmonics of that frequency up to infinity. The multiple bands (tines) of the Harmonic Reject filter will attenuate the fundamental as well as all of the harmonic by-products within the audio spectrum. Should you encounter wide-band line frequency related noise that is asymmetrical, it can produce some even harmonics.

An example of noise containing even harmonics would be that produced by an unsymmetrical sine, trapezoidal or square waveform. The Harmonic Reject filter can be placed in a mode in which it will attenuate the "evens" rather than the "odds" if you should encounter such noise. Before using the Harmonic Reject Filter, it is often useful to identify the fundamental frequency of the noise signal by using the Spectrum Analyzer found under the "View" menu.

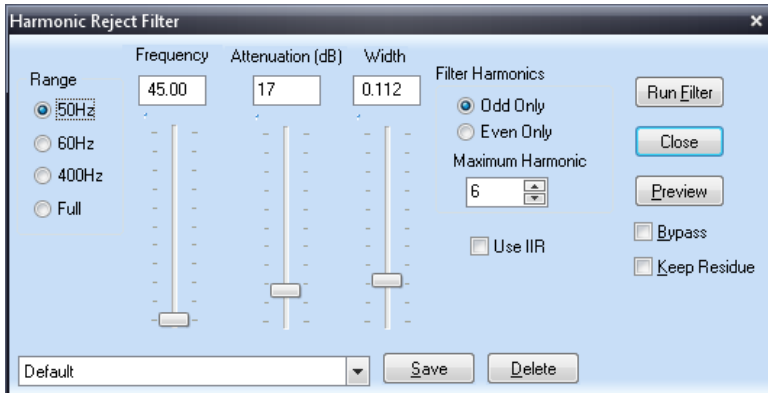


Figure 60 - The Harmonic Reject Filter

The following is a summary of the control parameters and range of adjustment provided for the Notch Filter:

1. Frequency (Fundamental): 20 - 5,000 Hz
2. Attenuation: 1 to 100 dB
3. Filter Harmonics:
 - A. Odd Only
 - B. Even Only
4. Maximum Harmonic (Number): 0 to 500
5. Width (or Q): 0.005 to 0.5 Octaves

6. Range: (This control optimized the resolution of the Frequency Control) - Choose between 50 Hz, 60 Hz, 400 Hz (aircraft) or Full Range
7. Use IIR Checkbox: Changes the mode from an FFT based system to a Resonant IIR Filter type

This filter incorporates the "Keep Residue" feature. This allows you to hear or keep only the noise component of the original signal. This feature is useful for "tuning" the filter to the maximum level of noise, so that when you actually run the filter with the "Keep Residue" feature turned off, the noise left behind will be minimized. Lastly, the keep residue function also allows you to produce "slot" filters, provided the slots that you need are harmonically related. See Appendix 1 for more details about slot filters.

Fine Tuning the Harmonic Reject Filter

Let's assume that you have a .wav file having a buzz which contains roughly a 60 Hz fundamental frequency. This can be determined by applying the spectrum analyzer to the file and making a measurement of the first spike in the series. Set the frequency of the Harmonic Reject filter range and also its frequency slider to 60. Start with a width of 0.2. Set the filter harmonics to 50 and check the "Keep Residue" mode. Next, click on the preview button and then point your mouse to the frequency slider. You should be hearing mostly buzz and not much signal in this mode. Now, use the up and down keys on your keyboard to move the frequency first above 60 Hz and then below 60 Hz one small increment at a time. Keep on adjusting the frequency in small increments until you hear the loudest buzz coming from your sound system. When you have found the optimal value, switch back to non-keep residue mode. You will hear the signal with the buzz substantially attenuated. The next thing to do is to adjust the Filter Harmonic number to the minimum value required to attenuate the buzz without affecting the quality of your audio signal. Lastly, adjust the width to the minimum value that will do the job for you so as to minimize damage to the primary signal of interest.

Sometimes, a better result is achieved using the IIR (resonant filter) technique. You can enter that mode via the IIR checkbox. This allows a much higher frequency resolution adjustment capability and so you

may be able to more closely hone in on the target signal of interest. However, when using the IIR method, try to minimize the number of harmonics to no more than what is needed to get the job done (Maximum Harmonic). The IIR method is much more math intensive and thus, slower, especially with high values of Maximum Harmonic settings.

Important Note:

For severe 50 or 60 Hz buzz situations, run at least two passes of the filter. The first pass should be run with a setting of 50 or 60 Hz, odd, and 500 harmonics and the second pass, run with a setting of 25 or 30 Hz, odd, and 500 harmonics. This sequence can be repeated more than once for further buzz reduction. Don't forget that you can stack up 2 or more of these filters in the Multi-Filter and apply them all at once.

Dynamic Noise Filter



(Analog Noise Filter)

This is a digital simulation of a dynamic analog filter. It is useful for dynamically attenuating "Hiss" from old record recordings or from old magnetic tape recordings. It performs better than a fixed Low-pass filter because it only attenuates high frequencies when there is no high frequency information present above the setting of its "threshold" adjustment. Sometimes this technique is referred to as "single-ended noise reduction." The Dynamic Noise filter's Low-pass corner frequency is frequency modulated by a rectified envelope signal that represents the amplitude of the signal content above a particular low-pass corner. So, normally, the bandwidth of this filter is limited until some high frequency content is measured by its high frequency detector. When this occurs, the bandwidth of the filter is opened up to allow the frequency of interest to pass through. When the high frequency signal diminishes again below a threshold value, the filter closes back down to a smaller bandwidth. The user has the ability to adjust a number of parameters with this filter, including noise threshold, filter frequency, attack time (the time constant associated with the signal whose job it is to increase the Low-pass filter frequency corner), release time (the time constant associated with the signal whose job it is to decrease the Low-pass filter frequency corner after a high frequency event has ceased) and HF (high frequency) gain. This

filter should only be used on recordings that contain little or no impulse noise, or on recordings that have already been processed through the Impulse Noise filter first to minimize unnecessary filter “breathing” effects.

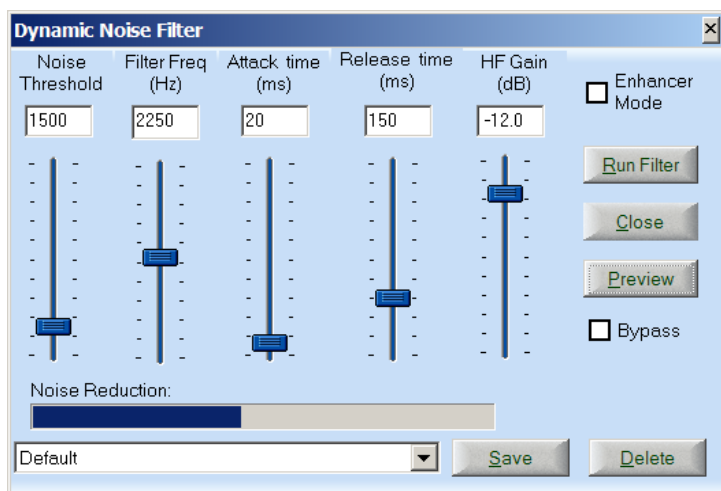


Figure 61 - The Dynamic Noise Filter

The Dynamic Noise Filter provides the following slider controls:

- **Noise Threshold**

The value of rectified and averaged High-pass signal Voltage above which the output of the Dynamic Noise Filter starts to raise its corner frequency. Moving the noise threshold slider control vertically raises its value. This control must be adjusted so that when there are no highs present in the source material, background "Hiss" is attenuated, but when "highs" (such as cymbal crashes or the pronunciation of the letter "S") the filter "opens up".

- **Filter Frequency**

This is the 1st order High-pass corner filter frequency which drives the Dynamic Noise Filter Detector / Rectifier / Attack & Release Time Constant circuitry. For modern reel-to-reel tapes, this parameter will be operated generally up in the 4 to 6 kHz range. For early 78s, it will be operated in the 1 to 3 kHz range. The Filter Frequency parameter

range is from 200 Hz to 19.99 kHz. Experimentation will be required to determine the best settings.

Important Note:

The frequency range of adjustment up to 19,999 Hz is only effective when utilizing a 44.1 kHz sampling rate or higher. At a 22.05 kHz sampling rate, the maximum effective frequency setting will be 10 kHz, and at an 11.025 kHz sampling rate, this value will drop to 5 kHz.

- **Attack Time**

The Attack Time slider adjusts the time constant (in milliseconds) associated with the rising edge of a High-pass signal envelope. Fast music will require smaller values of attack time compared to slow music. The range of adjustment for the Attack Time parameter is 1 to 300 milliseconds.

- **Release Time**

The Release Time slider adjusts the time constant (in milliseconds) associated with the falling edge of a High-pass signal envelope. This parameter will also require smaller values for fast music compared to the requirements of slow music. Also, the release time will almost always be set to a period of time greater than the Attack time for the algorithm to sound natural. The range of adjustment for the Release Time parameter is 1 to 500 milliseconds.

- **Gain**

Gain controls the amount of dynamic High-pass filter signal that is summed back into the output of the filter. This allows you to obtain upward or downward expansion of the high frequency portion of the audio spectrum. The "neutral" setting for this would be 0 dB that represents no expansion or compression. Setting this value greater than 0 dB will produce a "Spectral Enhancer" function. Values of 0 dB and lower produce a Single Ended Noise Reduction function used to de-Hiss a sound source. This control is calibrated in dB.

Important Note:

The controls can be adjusted "live" when the preview mode button is clicked.

Dynamic Noise Filter Operating Procedure (Tutorial)

Most of the parameter settings for this filter will vary considerably depending on the content of your particular .wav file. You will have to experiment to determine the values most to your liking. The values used below in the Procedure Example will get you started.

1. Highlight the portion of your .wav file on which you desire to apply the Dynamic Noise Filter. (You may choose to highlight the entire file or any portion thereof.)
2. Click on the "Filter Menu" with the left mouse button.
3. Click on "Dynamic Noise Filter."
4. Set the Noise Threshold slider all the way down; this is the minimum threshold position of the slider control.
5. Set the Filter Frequency to around 1.5 kHz.
6. Set the Attack Time to about 5 mSec. (Unless you are attempting to obtain some sort of special effect, the Release Time should always be set to a value greater than or equal to the Attack Time.)
7. Set the Release Time to around 50 mSec.
8. Set the Gain Control to 0 dB. (This control should only be set to higher numbers if a "Spectral Enhancement" effect is desired.) This will modify the "effective bandwidth" of your recording by incrementally amplifying the high end of the spectrum when there are enough highs present to trip the detector. This can be used to increase the "presence" of a recording, or to enhance the sound of a vocalist. If greater noise reduction is desired set the gain control to higher negative values.
9. Click on "Preview."

When the filter is operating properly, "Hiss" will be reduced, but when there is high frequency content on the recording, the filter should "open up" and pass through the "highs". If the Threshold is set too high, the filter will never open up, and the .wav file will sound "dull" although "Hiss" may be reduced. If the Threshold is set to low, the filter will always be opened up to full bandwidth, and there will be no noise reduction. If the Attack time is set too long, there will be a delay heard before the filter changes bandwidth on musical high frequency transients such as cymbal crashes. If the Release time is set too long,

there will be a residual "Hiss" left behind after a high frequency musical event, which will decay out, but too slowly.

When you determine the best setting of the controls for your particular .wav file, click Run filter. When the filter has completed its operation, the results will appear in the Destination Workspace.

Low Pass, Band Pass and High Pass IIR Filter Sub-Menu

The IIR based Low Pass, Band Pass and High Pass Filters are all in a sub-menu called "LP, BP, HP Filters".

Low Pass Filter



This filter is called a Low-pass filter because it only passes through signals that are lower than its set corner frequency. It attenuates high frequency signals above the corner frequency.

The effect can be similar to turning down the treble control on a home stereo except that the Low-pass filter is much more flexible. This filter can be somewhat useful for reducing hiss in a recording, but care must be taken not to reduce the "presence" of a recording by eliminating too much of the high end musical content at the same time.

Low Pass Filter with Chebyshev or Butterworth Response with up to 4th Order Slope

This type of filter is most useful when a recording does not contain any musical information above a certain frequency, and you wish to attenuate that high frequency noise that would otherwise be present.

This is a digital simulation of a conventional Low-pass filter. It is created using an Infinite Impulse Response (IIR) algorithm having a Butterworth or Chebyshev characteristic (for the higher order slopes). Frequencies below the "corner frequency" are passed through to the output, and frequencies above the corner are attenuated. The degree to which higher frequencies are attenuated is determined by the slope (order) of the filter. Four slopes are provided. They are 6dB / Octave, 12 dB / Octave, 18 dB / Octave and 24 dB / Octave. The higher the slope, the more attenuation will occur to frequencies above the corner frequency. The corner frequency is the frequency that you choose, and it is defined as the frequency at which the signal has been attenuated by

3 dB. This filter can be somewhat useful for reducing hiss in a recording, but care must be taken not to reduce the presence of a recording by eliminating too much of the high end musical content at the same time. When used selectively, this filter can also be used either to "De-Ess" an overly sibilant vocal, or reduce harsh harmonic distortion products that may have resulted from occasional master recording overloading (clipping). For forensic recordings, this filter can be used to remove most sounds whose frequencies are above the speech spectrum by setting the corner frequency to around 3,000 Hz.

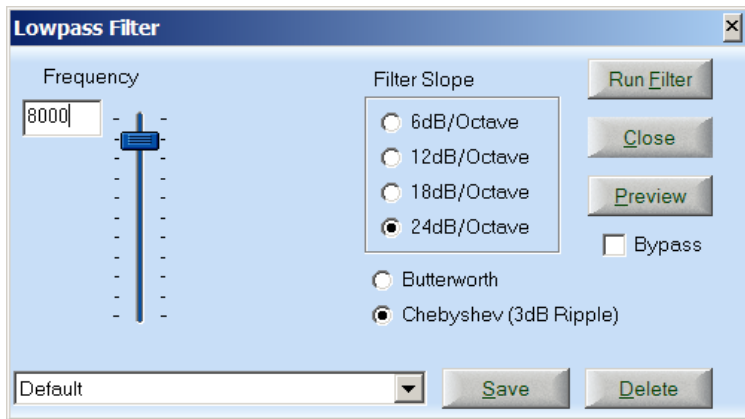


Figure 62 - The Low Pass Filter

The higher order (12, 18, & 24 dB / Octave) Low-pass filters are of the Butterworth or Chebyshev types, depending on your choice.

The following is a summary of the control parameters and range of adjustment provided for the Low-pass Filter:

- A. Frequency: 5 - 19,999 Hz
- B. Filter Slope: 6, 12, 18 & 24dB / Octave
- C. Preview Mode Button: On / Off (The slider control can be adjusted "live" when preview mode is on.)
- D. Filter Type: Choice of Butterworth or Chebyshev

Important Note: The frequency range of adjustment up to 19,999 Hz is only effective when utilizing a 44.1 kHz sampling rate. At a 22.05 kHz sampling rate, the maximum effective frequency setting will be 10 kHz, and at an 11.025 kHz sampling rate, this value will drop to 5 kHz.

Low pass Filter Operating Procedure (Tutorial)

1. Highlight the portion of your .wav file on which you desire to apply the Low-pass filter. (You may choose to highlight the entire file or any portion thereof.)
2. Click on the "Filter Menu" with the left mouse button.
3. Click on "Low-pass."
4. Choose the "Frequency" above which you desire to attenuate all signals, utilizing the left mouse button in conjunction with the "Frequency" slider control. When the control is all the way down, the setting will be 5 Hz, and when it is all the way up, the setting will be 19,999 kHz. (Useful settings will usually fall somewhere within the 3 kHz to 15 kHz range, depending on the source material and the goals of the audio restoration.) If you desire finer frequency resolution, you may also use direct numeric entry of the value.
5. Choose the Filter "Slope" which you desire. Click on 6 dB / Octave, 12 dB / Octave, 18 dB / Octave or 24 dB / Octave. The steeper the slope, the higher will be the degree of attenuation of all frequencies above the "Frequency" setting.
6. If you desire to hear the results of your filter settings before creating a new "Destination" file, click on "preview."
7. After a short delay, you will hear the effect of the settings that you have chosen. (The system may seem to stutter if your computer is too slow to keep up with the algorithm in real time. However, this repeating pattern will not be present in the final Destination processing of the filter.)
8. As the filter is running in either Preview mode or Run mode (Destination File Mode), you will see a dialog box which indicates the "% Done" of the filter algorithm on the selected portion of the Source .wav file. Also, at the top of the Dialog box you will see indicated the "Total Samples to Process:"
9. Keep adjusting the Frequency and Slope parameters, and testing the various settings using the "Preview" mode until you are satisfied with the results.

10. When you are satisfied with a group of settings, you will be done with Preview mode.
11. Click on “Run”, and the filter will process your Source .wav file through the filter algorithm, and create a Destination .wav file containing the output of the filter.
12. When this process is complete, you will see the Destination File become highlighted in Yellow, at the same time that the Source File becomes unselected.

Band Pass Filter



Band-pass filters are essentially a combination of a Low-pass filter and a High-pass filter. It attenuates both the high frequency and the low frequency portions of the audio spectrum. It is useful where the recording contains extraneous noise in the low frequency region such as rumble or thumps, and high frequency noise such as hiss. This filter can also be very useful for improving the intelligibility of audio recordings, especially speech, by eliminating the unnecessary portion of the audio spectrum that is not used by speech frequencies to carry useful information to the listener.

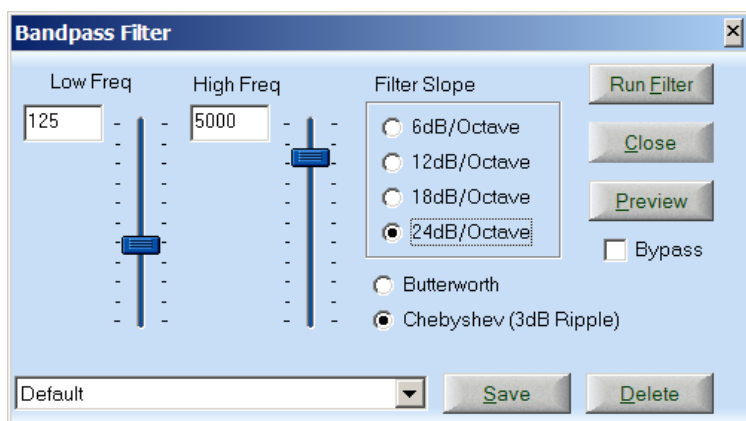


Figure 63 - The Band Pass Filter

Band Pass Filter with Chebyshev or Butterworth Response with up to 4th Order Slope

This is a digital simulation (IIR based) of a conventional analog Band-pass filter having a Butterworth or Chebyshev response when set to the steeper slope values. Band-pass filters are passive to frequencies within the Band-pass region, but they attenuate frequencies above and below the two corner frequencies. Band-pass filters have both an upper and a lower corner frequency, and like the Low-pass and the High-pass filter, the corner frequencies are defined as the frequencies at which the signals either above the upper corner or below the lower corner are attenuated by 3 dB. Four slopes are provided for the Band-pass Filter, just like the Low-pass and the High-pass. They are 6 dB / Octave, 12 dB / Octave, and 18 dB / Octave and 24 dB / Octave. This filter can be very useful for improving the intelligibility of audio recordings, especially speech, by eliminating the unnecessary portion of the audio spectrum that is not generally occupied by vocal frequencies used to carry useful information to the listener. The Forensics Menu "Brick Wall" filter has a much steeper version of this filter for dealing with extreme cases of out-of-band noise that needs to be eliminated.

Note:

The higher order (12, 18, & 24 dB / Octave) Band-pass filters are of the Butterworth or Chebyshev type depending on your choice.

Special effects can be produced with the Band-pass filter. These special effects can be useful when producing movies or stage plays or shows and a particular sound producing device and its environment needs to be accurately reproduced through the "House" P. A. System. Here are a few examples:

Simulation	Low Freq. Control	High Freq. Control	Slope
1930's Vintage Table Top Radio:	830 Hz	2000 Hz	18 dB / Octave
Modern cheap Table Top Radio:	265 Hz	6100 Hz	18 dB / Octave
Loud "Walkman" personal stereo as heard by person nearby:	3650 Hz	9800 Hz	18 dB / Octave

Modern Stereo System as heard from the next room:	95 Hz	4100 Hz	12 dB / Octave
1950's Vintage Juke Box:	30 Hz	2700 Hz	12 dB / Octave
AM Transistor Pocket Radio:	1395 Hz	2110 Hz	18 dB / Octave
Telephone Receiver sound from "off the hook":	2700 Hz	2895 Hz	18 dB / Octave
Night Club Band as heard from Parking Lot:	85 Hz	240 Hz	12 dB / Octave
Olde Acoustic Phonograph:	870 Hz	2390 Hz	18 dB / Octave
Public Address System at Outdoor Event:	300 Hz	3000 Hz	12 dB / Octave
Modern High End Audio System:	15 Hz	19,999 Hz	6 dB / Octave
Bandpass Filter Response Limits:	5 Hz	19,999 Hz	-

You can create your own simulations of sound devices and acoustic environments through experimentation with the Band-pass filter parameters. Using the above simulations, in conjunction with the DC8/DC FORENSICS reverb, you can further enhance various acoustical environments. Once you discover the appropriate values, write them down or store them as presets for future reference.

Cascading this filter (using the Multi-Filter) with others like the Virtual Valve Amplifier to add distortion, and the Reverb to add room acoustical effects can embellish these sound simulations.

The Band-pass filter can also be used as a tool to determine if any useful audio information exists in a particular portion of the audio spectrum; it becomes sort of an audible wave analyzer when used in this manner. For more information on this mode of operation, refer to the "Using DC8/DC FORENSICS as an Audio Waveform Analyzer" portion of the "How Do I" section of this manual or go to the section which explains the operation of the Spectrum Analyzer.

The following is a summary of the control parameters and range of adjustment provided for the Band-pass Filter:

- Low Frequency: 5 - 19,999 Hz.
- High Frequency: 5 - 19,999 Hz.
- Filter Slopes: 6, 12, 18, & 24 dB / Octave.

- Preview Mode Button: On / Off (The slider controls can be adjusted "live" when preview mode is on.)
- Filter Type: Choice of Butterworth or Chebyshev

Note: The frequency range of adjustment up to 19,999 Hz is only effective when utilizing a 44.1 kHz sampling rate. At a 22.05 kHz sampling rate, the maximum effective frequency setting will be 10 kHz, and at an 11.025 kHz sampling rate, this value will drop to 5 kHz.

Band-pass Filter Operating Procedure (Tutorial)

1. Highlight the portion of your .wav file on which you desire to apply the Band-pass filter. (You may choose to highlight the entire file or any portion thereof.)
2. Click on the "Filter Menu" with the left mouse button.
3. Click on "Band-pass."
4. Make an initial determination of what band of frequencies you desire to pass through the Band-pass filter.
5. Utilizing the right mouse button in conjunction with the Low Frequency slider control, select the lower corner frequency of the range that you have chosen. (The range for this control is 5 Hz to 19,999 kHz)
6. Utilizing the right mouse button in conjunction with the High Frequency slider control, select the upper corner frequency of the range that you have chosen. (The range for this control is 5 Hz to 19,999 kHz)
7. If you desire finer frequency resolution for either the lower or the upper corner frequency, you may use direct numeric entry, instead of the slider controls.
8. Choose the Filter "Slope" which you desire. This slope will symmetrically affect both the upper and lower corner roll-off rates. Click on either 6 dB / Octave, 12 dB / Octave, 18 dB / Octave or 24 dB / Octave. The steeper the slope, the higher will be the degree of attenuation of all frequencies outside of the selected pass band range.
9. If you desire to hear the results of your filter settings before creating a new "Destination" file, click on "preview."
10. After a short delay, you will hear the effect of the settings that you have chosen. (The system may seem to stutter if your computer is too slow to keep up with the algorithm in real-

time. However, this repeating pattern will not be present in the final Destination processing of the filter.)

11. Keep adjusting the Low Frequency and High Frequency sliders as well as the Slope parameters until you achieve the effect you desire and are satisfied with the results.
12. When you are satisfied with a group of settings, you will no longer need to invoke the Preview function.
13. Click on "Run", and the filter will process your Source Wave file through the Band-pass Filter algorithm, and create a Destination .wav file containing the output of the filter.
14. When this process is complete, you will see the Destination File become highlighted in Yellow, at the same time that the Source File becomes unselected.
15. Click on "Close"

Note 1: If the low frequency control is set to a higher frequency than the high frequency control setting, a "no pass" filter will be created. This is of little useful value, but is allowable by DC8/DC FORENSICS.

Note 2: You can select Chebyshev rather than Butterworth if you want steeper filter response at the expense of some ripple within the filter pass-band.

High Pass Filter



A High-pass filter only passes signals that are above or "higher" than the corner frequency. It reduces the level of low frequency signals that are below the corner frequency. The effect can be similar to turning down the bass control on a home stereo. This filter is very useful for reducing turntable rumble, muddiness, and any other extraneous low frequency noise in a recording.

High Pass Filter with Chebyshev or Butterworth Response with up to 4th Order Slope

This is a digital simulation (IIR based) of a conventional analog High-pass filter having a Butterworth or Chebyshev response (for the higher order slopes). This filter attenuates frequencies below the High-pass corner frequency. Just like the Band-pass and the Low-pass filter, this filter has four slopes available. They are 6 dB / Octave, 12 dB /

Octave, 18 dB / Octave and 24 dB / Octave. This filter is very useful for reducing the effects of turntable rumble, or microphone seismic effects from creating "muddiness" on an audio recording. It can also be used to eliminate any dc (fixed) offset that may have developed on a .wav file (a preset is available which performs this function). To reduce turntable rumble, start with a setting of 60 Hz and 18 dB / Octave, and then adjust the parameters until you are satisfied. This filter is also useful when selectively applied for reducing microphone "P" popping effects on the vocal track of multi-track recordings wherein an adequate windscreen had not been utilized in the session. Effective settings to attenuate "P" popping typically are 120 Hz with a slope of 18 dB / Octave (selectively applied to the event).

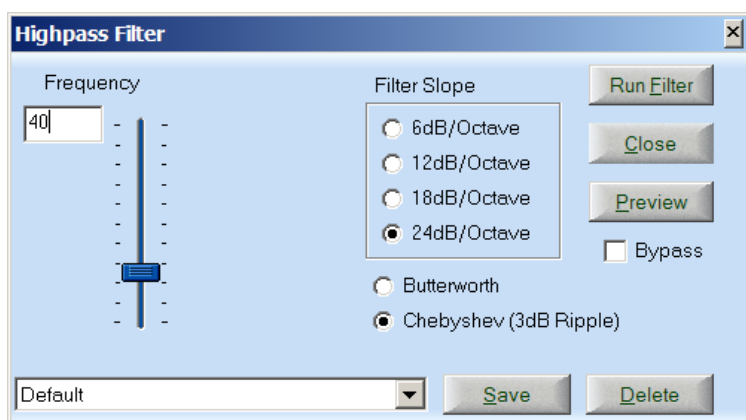


Figure 64 - The High Pass Filter

Note: The higher order (12, 18, & 24 dB / Octave) High-pass filters are of the Butterworth or Chebyshev types, depending upon your selection.

The following is a summary of the control parameters and range of adjustment provided for the High-pass Filter:

- **Frequency:** 5 - 19,999 Hz.
- **Slope:** 6, 12, 18, 24 dB / Octave

- **Preview Mode Button:** On / Off (The slider controls can be adjusted "live" when the preview mode button is activated.)
- **Filter Type:** Choice of Butterworth or Chebyshev

Note: The frequency range of adjustment up to 19,999 Hz is only effective when utilizing a 44.1 kHz sampling rate. At a 22.05 kHz sampling rate, the maximum effective frequency setting will be 10 kHz, and at an 11.025 kHz sampling rate, this value will drop to 5 kHz.

High-pass Filter Operating Procedure (Tutorial)

1. Highlight the portion of your .wav file on which you desire to apply the High-pass filter. (You may choose to highlight the entire file or any portion thereof.)
2. Click on the "High Pass Filter". (Filter Menu)
3. Choose the "Frequency" below which you desire to attenuate all signals, utilizing the right mouse button in conjunction with the "Frequency" slider control. When the control is all the way down, the setting will be 5 Hz, and when it is all the way up, the setting will be 20 kHz. (Useful settings will usually fall somewhere within the 15 Hz to 500 Hz range, depending on the goals of the audio restoration process.) If you desire finer frequency resolution, you may also use direct numeric entry of the value.
4. Choose the Filter "Slope" which you desire. Click on either 6 dB / Octave, 12 dB / Octave, or 18 dB / Octave or 24 dB / Octave. The steeper the slope, the higher will be the degree of attenuation of all frequencies below the "Frequency" setting.
5. Click on Preview to audition the results before processing.
6. After a short delay, you will hear the effect of the settings that you have chosen. (The system may seem to stutter if your computer is too slow to keep up with the algorithm in real time. However, this repeating pattern will not be present in the final Destination processing of the filter.)
7. As the filter is running in either preview mode or normal mode (Destination File Mode), you will see a dialog box that indicates the "% Done" of the filter algorithm on the selected portion of the Source .wav file. Also, at the top of the Dialog box you will see indicated the "Total Samples to Process:"

8. Keep adjusting the Frequency and Slope parameters, and testing the various settings using the "Preview" mode until you are satisfied with the results.
9. When you are satisfied, click on Run, and the filter will process your Source .wav file through the filter algorithm, and create a Destination .wav file containing the output of the filter.
10. When this process is complete, you will see the Destination File become highlighted in Yellow, at the same time that the Source File becomes unselected.

Note: For steeper filter response at the expense of pass-band ripple, choose Chebyshev rather than Butterworth response.

Removing DC-Offsets with the High-pass Filter (Tutorial)

The High-pass Filter can be used to remove DC-Offsets from .wav files. To do so, set the High-pass Filter to 10 Hz and 6 dB per octave, and run it on the .wav file needing correction.

Notch Filter



A notch filter attenuates signals that are near its center frequency setting. The degree to which it attenuates frequencies near the center frequency is determined by the bandwidth setting. This filter is useful for removing 50 or 60 Hz hums from a recording. It is also useful for decreasing any sound system acoustic feedback that may be found on some live recordings. A "Slot" filter is also provided within the Notch filter menu item for Forensics applications. Multiple slots can be constructed using the DC8/DC FORENSICS Multi-Filter.

This is a digital simulation (IIR based) of a second order notch / slot filter. It attenuates all frequencies near its center frequency setting. The degree to which it attenuates frequencies adjacent to the center frequency is determined by the bandwidth setting. This filter is useful for removing 50 or 60 Hz hum from a recording (or harmonics thereof). It is also useful for decreasing any sound system acoustic feedback that may be found on some live recordings. It can be used to attenuate the heterodyning "whistle" which is sometimes heard on AM broadcast

radio reception. Some audio restoration engineers also use this filter to remove some "Hiss" from recordings. For this application, the filter's center frequency is set somewhere in the 8 to 12 kHz range, with a bandwidth of 0.25 Octave or less. Experimentation is the only way to determine its effectiveness in minimizing "Hiss" from your particular source material. Also, it is important to note that this method is not the most effective for "Hiss" removal. Instead, consider using either the Continuous Noise filter or the Dynamic Noise filter.

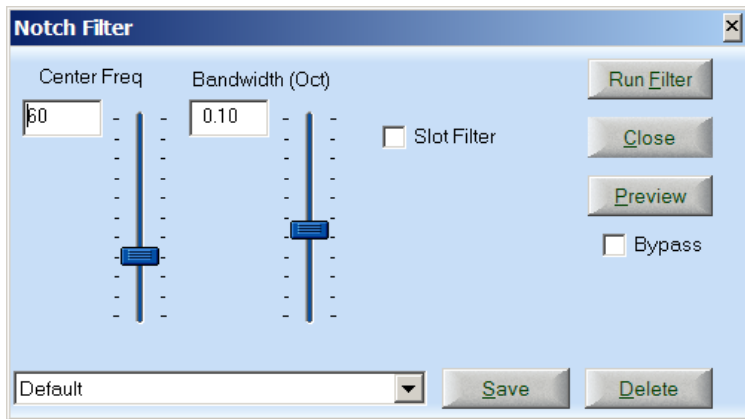


Figure 65 - The Notch Filter

The following is a summary of the control parameters and range of adjustment provided for the Notch Filter:

- **Center Frequency:** 5 - 19,999 Hz
- **Bandwidth:** 0.01 Octaves to 1.99 Octaves
- **Preview Mode Button:** On / Off (The slider controls can be adjusted "live" when the preview mode is on.)

This filter incorporates the "slot filter" feature. Essentially, a slot filter produces the inverse response of a notch filter. This allows you to hear or keep only the residual component of the original signal. This feature is useful for "tuning" the notch filter to the maximum level of noise, so that when you actually run the notch filter with the "slot filter" feature turned off, the noise left behind will be minimized. The slot filter function is also useful in Forensics applications, wherein one is

interested in isolating a very particular sound that exists in a very specific and narrow frequency band. If multiple slots are required, use the Multi-Filter with multiple notch/slot filters in the chain, or consider using the Harmonic Reject filter in "Keep Residue" mode, provided the slots, which are required, are harmonically related.

Note: The frequency range of adjustment up to 19,999 Hz is only effective when utilizing a 44.1 kHz sampling rate. At a 22.05 kHz sampling rate, the maximum effective frequency setting will be 10 kHz, and at an 11.025 kHz sampling rate, this value will drop to 5 kHz.

Notch Filter Procedure (Tutorial)

1. Identify the frequency that you desire to reject from the .wav file recording. The determination of the frequency of interest can be simplified by using the Band-pass filter as an Audible Spectrum Analyzer.
2. Highlight the portion of your .wav file on which you desire to apply the Notch filter. (You may choose to highlight the entire file or any portion thereof.)
3. Click on "Notch."
4. Choose the Center Frequency based on the "problem" frequency that you have observed and desire to reject. The Center Frequency range is from 5 Hz. to 19,999 kHz. (The highest useful frequency is around 15 kHz.) If you desire the finest degree of frequency resolution possible, use direct numeric entry rather than the use of the slider controls.
5. Choose the Bandwidth that you find to be the most effective in eliminating the desired frequency. The bandwidth control is calibrated in Octaves, and has a range from 0.01 Octaves to 1.99 Octaves. You should choose the smallest possible bandwidth that still accomplishes the job of rejecting the troublesome frequency. Otherwise, you will start eliminating useful information from your recording. Generally, useful ranges for Bandwidth will be in the 0.5 Octave to 0.1 Octave range.
6. If you desire to hear the results of your filter settings before creating a new "Destination File, click on "preview."
7. After a short delay, you will hear the effect of the settings that you have chosen. (The system may seem to stutter if you

computer is too slow to keep up with the algorithm in real time. However, this repeating pattern will not be present in the final Destination processing of the filter.)

8. As the filter is running in either preview mode or normal mode (Destination File Mode), you will see a dialog box that indicates the "% Done" of the filter algorithm on the selected portion of the Source .wav file. Also, at the top of the Dialog box you will see indicated the "Total Samples to Process:"
9. Keep adjusting the Frequency and Slope parameters, and testing the various settings using the "Preview" mode until you are satisfied with the results.
10. When you are satisfied with a group of settings, you will no longer need to use Preview mode.
11. Click on Run, and the filter will process your Source .wav file through the filter algorithm, and create a Destination .wav file containing the output of the filter.
12. When this process is complete, you will see the Destination File become highlighted in yellow, at the same time that the Source File becomes unselected.
13. Click on "Close."

Note: If only a small section(s) of your .wav file is in need of Notch filtering, as is often the case when acoustic feedback is encountered on a live recording, you can use "sync mode" and just filter the sector which contains the noise which your are attempting to reduce. "Sync mode" can be selected under the "View Menu."

Median Filter



The Median Filter can be used to substantially reduce "crackle" (small impulse noise) from a recording. Use a sample setting of 3 to 7 for this application. A "weighting" control is also provided, which affects the "timbre" of the processed sound.

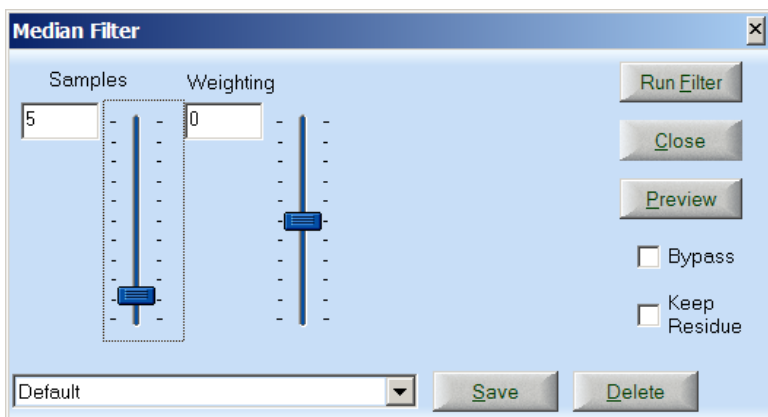


Figure 66 - The Median Filter

There is no analog equivalent to the Median Filter. This filter defines a window of samples, and for that window, determines which sample is the median value within the grouping (median meaning middle value). That value is the one that is passed along to the destination file, and then the window moves over 1 sample and re-evaluates the median, again passing the new median value to the destination file. This filter is useful for improving the intelligibility of severely distorted signals and it is also useful for pulling signals out of a very poor signal-to-noise ratios situation (pulling signals out of the mud). It is somewhat similar in sound performance to a high-order Low-pass filter. (The median value of a string of sorted numbers is the one in the middle of the string. In other words, if you have seven sorted numbers, the fourth number will be the median value.) The DC8/DC FORENSICS Median Filter allows you to choose the number of samples over which the median value is determined by the algorithm. The range is from 3 to 20 samples. The higher that you set this value, the greater will be the attenuation of high frequency signals and/or noise. Also, the higher the samples setting, the longer will be the processing time required for the filter. The most useful settings will generally be found to be from 3 to 7 samples. Outside of that range, you may hear a significant degradation of the top end of your recording, depending on its bandwidth. Even at 7 samples, you will notice some "fuzziness" intermodulating with the upper end of the spectrum on some recordings. So always start with 3 to 5 samples when using the Median filter, and choose the smallest value that produces an effective de-crackling result.

It is also important to note that the higher the number of samples selected, the longer it will take your computer to calculate the Median values to process your .wav file. At some high settings, your system may not be able to process the file in "real-time" due to the demands it places on your CPU.

Weighting Function

A "Weighting" function is provided with this filter that shifts the position on the number line as to which value will be chosen as the "Median." This feature is particularly useful in forensics audio application wherein extremely inarticulate speech needs clarification. For example, the absence of consonant or sibilant sounds can render a recording undecipherable. Recording bandwidth limitations can severely distort or eliminate the "hissing-consonants" which render speech understandable - - a fact that has been known since the earliest ventures by Thomas Edison into the field of recorded sound. With the use of the Median filter in conjunction with the weighting control, this problem can often be corrected. The weighting control will essentially affect the "timbre" of the processed sound. Trial and error will determine the best combination of "Samples" and "Weighting" to cure a particular forensics sound problem.

The following is a summary of the control parameters and range of adjustment provided for the Median Filter:

- **Samples:** 3 - 20
- **Preview Mode Button:** On / Off (The slider control can be adjusted "live" when the preview mode is on.)
- **Weighting:** +/- 100 from the neutral value of 0 (0 provides no weighting to the Median value calculated by the filter)
- **Bypass Button:** Allows you to compare between the filtered and the raw signal.
- **Keep Residue:** Allows you to hear what the Filter is removing from the raw signal.

Note: De-Crackling will generally be best accomplished with a "Samples" setting around 3. Intelligibility improvement of extremely distorted or garbled voice recordings will generally be accomplished with "Samples" settings anywhere between 5 through 17. You will have to experiment to determine the best settings for solving a particular problem.

Median Filter Operating Procedure (Tutorial)

1. Highlight the portion of your .wav file on which you desire to apply the Median Filter.
2. Click on the "Median Filter" (Filter Menu)
3. Choose the number of samples over which you desire the median calculation to be performed. The higher the number of samples selected, the greater will be the attenuation of the higher frequency portion of the audio spectrum. You can choose any integer value from 3 to 20 samples. (The most useful values will be found in the 3 to 7 samples range.) The higher the chosen number of samples, the longer will be the process time requirement for the algorithm. Changes in value are accomplished utilizing the slider control.
4. If you desire to hear the results of your filter settings before creating a new "Destination" file, click on "preview."
5. You will hear the effect of the calculation of the median value over the chosen number of "samples."
6. As the filter is running in either preview mode or normal mode (Destination File Mode), you will see a dialog box that indicates the "% Done" of the filter algorithm on the selected portion of the Source .wav file. Also, at the top of the Dialog box you will see indicated the "Total Samples to Process:"
7. Keep adjusting the number of samples until you achieve your desired effect.
8. When you are satisfied with a setting, click on "Run", and the filter will process your source .wav file through the filter algorithm, and create a Destination .wav file containing the output of the filter.
9. When this process is complete, you will see the Destination File become highlighted in Yellow, at the same time that the Source File becomes unselected.

10. Click on "Close".

Note: The “Weighting” control is used to affect the Timbre of the resultant sound of the Median filter.

Averaging Filter



This filter sounds similar to that of a Low-pass filter, although it is somewhat more effective than a Low-pass filter in reducing not only "Hiss" but also "Crackle" from a sound source. It is most effective on limited bandwidth sources such as old acoustic recordings made before 1925. This filter is also useful for improving the intelligibility of highly garbled voice communications recordings.

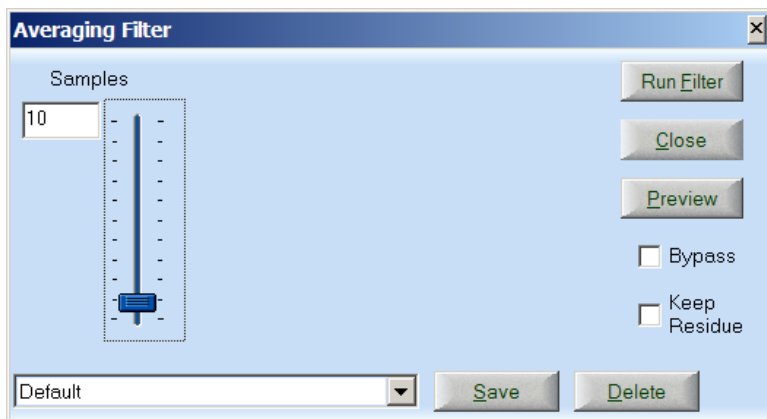


Figure 67 - The Averaging Filter

This is another filter that has no analog equivalent. Its interface to the operator is similar to the Median Filter, with the difference being that instead of calculating the median value of a sample window to pass into the Destination workspace, the average value of a group of samples is passed through. You select the number of samples on which the average value is calculated with a slider control in its dialog box. The greater the number of samples, the higher the degree of smoothing effect on the waveform. The higher that you set the degree of smoothing, the greater will be the loss in the higher end of the audio frequency spectrum.

The following is a summary of the control parameters and the range of adjustment provided for the Average Filter:

- **Samples:** 2 - 100
- **Preview Mode Button:** On / Off (The slider control can be adjusted "live" when the preview mode is being used.)
- **Bypass Button:** Allows you to compare between the filtered and the raw signal.
- **Keep Residue:** Allows you to hear what the Filter is removing from the raw signal.

Average Filter Operating Procedure (Tutorial)

1. Highlight the portion of your .wav file on which you desire to apply the Average Filter. (You may choose to highlight the entire file or any portion thereof.)
2. Click on the "Filter Menu" with the left mouse button.
3. Click on "Average."
4. Choose the number of Samples over which you desire the moving average calculation to be performed. The higher the number of samples chosen, the greater will be the attenuation of the higher frequency portion of the audio spectrum. You can choose any integer value from 2 to 100 samples. The higher the number of samples selected, the longer will be the processing time requirement for the algorithm. This selection is accomplished utilizing the slider control.
5. If you desire to hear the results of your filter settings before creating a new "Destination" file, click on "preview."
6. You will hear the effect of the averaging over the chosen value of "samples."
7. As the filter is running in either preview mode or normal mode (Destination File Mode), you will see a dialog box that indicates the "**Percent Done**" of the

filter algorithm on the selected portion of the Source .wav file. Also, at the top of the Dialog box you will see indicated the **"Total samples to process:"**

8. Keep adjusting the number of samples until you achieve your desired effect.
9. When you are satisfied with a setting, you will no longer use the Preview Mode button.
10. Click on "Run", and the filter will process your source .wav file through the filter algorithm, and create a Destination .wav file containing the output of the filter.
11. When this process is complete, you will see the Destination File become highlighted in Yellow, at the same time that the Source File becomes unselected.
12. Click on "Close".

The 10 Band Graphic Equalizer



The 10 Band Graphic Equalizer is a familiar filter that acts like an expanded tone control. The audio spectrum is broken into 10 bands, each being one octave wide. Each band's gain (volume) can be independently adjusted to achieve the desired audio result. This filter is useful for tonal shaping of the finished audio product or to enhance the bass or treble of a recording. It is also useful for improving the intelligibility of recordings or "Bringing Out" a particular instrument or vocal.

The Graphical Equalizer is the digital equivalent (which is IIR based) to the analog Graphical Equalizer found in many sound systems. Its primary advantage is that it can be applied to a .wav file without having to resort to adding an analog step in your sound restoration process. This results in decreased noise and distortion on your final product. The equalizer has ten bands containing the following center frequencies:

31 Hz, 62 Hz, 125 Hz, 250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz

8 kHz, 16 kHz

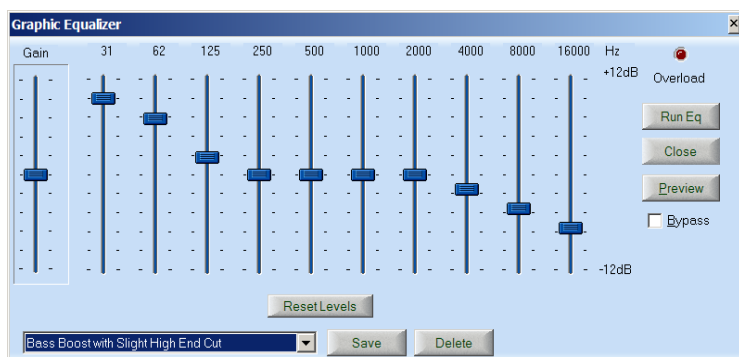


Figure 68 - The 10 Band Graphic Equalizer

The amplification and attenuation range for each band is + / - 12 dB. An additional feature is provided to allow a "shelf" function (this is a "pull out" in the frequency response curve, sometimes referred to as the adding of a "zero" in order to form a pole-zero pair) below 31 Hz and above 16000 Hz. A "Reset Levels" feature is provided. Clicking on "Reset Levels" will return all of the Graphical Equalizer slider controls to their 0 dB position. Since the graphic equalizer can add gain to your signal, a latching overload indicator is provided. If, at any time during the processing of a file through the graphic equalizer, the output signal exceeds the dynamic range of the system, the indicator will turn red and latch until the filter is re-run.

Note 1: The top equalizer band of 16,000 Hz is only effective when using a sample rate of 44.1 kHz or higher; it becomes ineffective at sampling rates of 22.05 kHz and 11.025 kHz. The 8,000 Hz band will also be rendered ineffective when using a sampling rate of only 11.025 kHz.

Note 2: The graphic equalizer controls can be adjusted "live" when the preview mode button is clicked.

Note 3: Since the graphic equalizer can actually increase the gain of the system, it is possible to produce clipping which will result in unpleasant distortion products to appear in the Destination Workspace. This usually occurs when excessively boosting the bass portion of the spectrum. The overload indicator will change from green to red, and latch in that condition if there has been an overload. The latch will be

reset, following a re-run of the algorithm, provided that the overload condition has been cleared by reducing gain in one or more bands.

Graphic Equalizer Operating Procedure (Tutorial)

1. Click on the Filter Menu.
2. Click on "Graphic Equalizer."
3. Using your Mouse, adjust the frequency band slider control up or downwards as desired. This can be accomplished by directly pointing the cursor with the mouse and depressing the left mouse button to move the control.
4. When the Slider control for a particular band is in its center position, the band is neither being attenuated nor amplified. Moving the slider upwards produces amplification of frequencies in the band up to 12 dB. Moving the slider downwards produces attenuation of frequencies in the band of up to 12 dB.

The 20 Band Graphic Equalizer



The 20 Band Graphic Equalizer is an IIR based extension of the 10 Band Graphic Equalizer. It exhibits twice the selectivity compared to the 10-band equalizer, since each band is calibrated with half its bandwidth. It is quite useful where greater selectivity is required in order to “tweak” your audio signal frequency distribution, but it is a little more difficult to use compared to the general purpose 10-band equalizer. Its amplification and attenuation range is +/- 12 dB as indicated by the numerical readouts located below each slider control. It has the following frequency band center frequency values:

22Hz, 31Hz, 42Hz, 63Hz, 88Hz, 125Hz, 177Hz, 250Hz, 350Hz,
500Hz, 710Hz, 1.0kHz, 1.4kHz, 2kHz, 2.8kHz, 4kHz, 5.6kHz, 8.0kHz,
11kHz, 16kHz

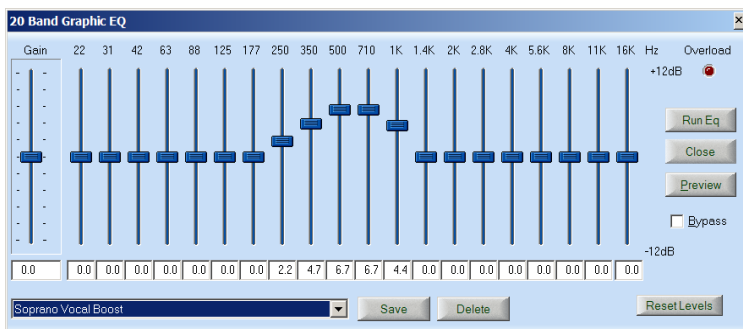


Figure 69 - The 20-Band Graphic EQ

In all other regards, the 20 Band Graphic Equalizer is operated in the same manner as the 10 Band Graphic Equalizer. *Please refer to the description of the Graphic Equalizer above for more details.*

Note: All bands above 45 % of the .wav files sampling frequency are disabled.

The 30 Band Graphic Equalizer (Forensics Version Only)



30 Band Graphic IIR-Based EQ

The 30 Band $1/3^{\text{rd}}$ Octave Graphic Equalizer is an IIR based extension of the 10 and 20 band Graphic Equalizers. It exhibits three times the selectivity compared to the 10-band equalizer, since each band is calibrated with $1/3^{\text{rd}}$ of its bandwidth. This is very useful in Forensics applications wherein signals in a relative narrow portion of the audio spectrum need to be amplified or attenuated. Its amplification and attenuation range is ± 12 dB as indicated by the numerical readouts located below each slider control. If this filter does not have sufficient selectivity for your application, then use the FFT based Spectral Filter, which has a choice of from 128 to as high as 32,000 bands. The 30 Band Graphic Equalizer has the following frequency band center frequency values:

25 Hz, 31 Hz, 40 Hz, 50 Hz, 62 Hz, 80 Hz, 100 Hz, 125 Hz, 160 Hz,
200 Hz, 250 Hz, 320 Hz, 400 Hz, 500 Hz, 640 Hz, 800 Hz, 1 kHz,
1.3 kHz, 1.6 kHz, 2 kHz, 2.5 kHz, 3.1 kHz, 4 kHz, 5 kHz,
6.2 kHz, 8 kHz, 10 kHz, 13 kHz, 16 kHz, 20 kHz

Note: All bands above 45 % of the .wav files sampling frequency are disabled.

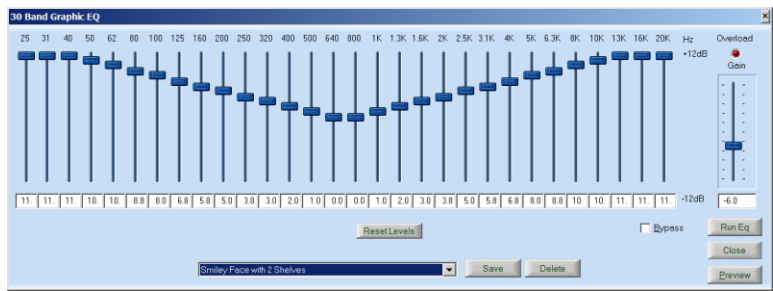


Figure 70 - The 30-Band Graphic EQ

Paraphrastic Equalizer



This unique equalizer combines the flexibility of a parametric equalizer with the ease of use associated with a 10 band graphic. The actual frequency response of the filter is graphically controlled and/or displayed. Virtually any frequency domain transfer function that you can dream of can be created with this filter. Many presets are provided to facilitate many unusual equalization situations.

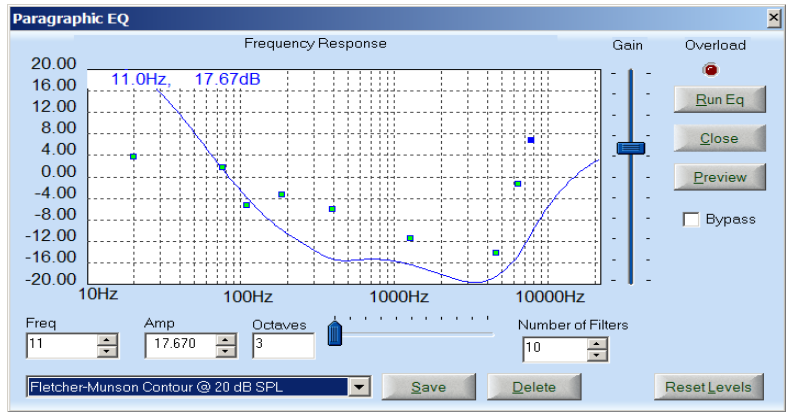


Figure 71 - The Paraphrastic Equalizer

The Paragraphic Equalizer is a unique form of a parametric equalizer employing IIR techniques. It combines the graphical representation of its transfer function (frequency response curve) with the versatility of a parametric equalizer. Additionally, it provides you with up to 10 bands of equalization. It differs from the graphic equalizer in that three parameters are adjustable for each band:

- **Frequency:** (Hertz - 10 to 20 kHz_
- **Amplitude:** (attenuation or gain - +/- 20 dB)
- **Octaves:** (Q or bandwidth - 0.05 to 3.0)

The following additional controls and displays are provided:

- **Number of filters:** (1 to 10)
- **Output Gain:** (+ / - 20 dB)
- **Reset Levels:** (Resets Paragraphic to zero dB and factory default frequencies)
- **Overload indicator:** (Illuminates when full-scale output {clipping} occurs.

The Parametric Equalizer displays its frequency domain transfer characteristic graphically. Therefore, it is referred to as the DC8/DC FORENSICS "Paragraphic" equalizer. It can be modified using your mouse by dragging the inflection point dots, and modifying the bandwidth using the octave control. You simply draw the shape of the response, which you desire, and the algorithm adjusts the parameters to match the response curve. Each band is represented by a single square "dot" on the graph. The "active" dot will be the larger one displayed. That is the dot for which the parameters are being displayed numerically on the control panel. You can use the mouse to drag any of the dots, which actually represents a frequency inflection point, wherever you wish. If you want to sharpen or widen the response of any inflection point, use the octave control to achieve the desired curve for a highlighted dot. It is often very useful to use the spectrum analyzer, found under the View menu, in conjunction with the Paragraphic Equalizer. You will then be able to see the exact effect that you are imposing on the .wav file signal.

Under the Settings menu, you will find a number of useful audio restoration functions, including the RIAA curves. Also, various inverse

RIAA curves with a variety of turnover frequencies are available. These features enable you to use a standard RIAA pre-amplifier to transfer acoustical and electrically recorded 78 RPM records to your hard drive, and re-compensate at another point in time, without having to purchase specialized hardware. Also, of interest is the family of Fletcher-Munson Equal Loudness Contours at different sound pressure levels. These can be used to compensate for the response of the human ear depending on the loudness level that you expect a particular audio piece to be auditioned.

Note 1:

As with all of the DC8/DC FORENSICS filters, sample theorem dictates useful bandwidth for the algorithms. The Paragraphic Equalizer will only have a useful bandwidth up to about 10 kHz with a 22.05 kHz sample rate, and about 5 kHz at 11.025 kHz.

Note 2:

Many of the Paragraphic Equalizer presets such as the RIAA, Reverse RIAA and NAB curves are defined over the entire audio spectrum consisting of at least 20 Hz to 20 kHz band spread values. Therefore, the use of any sampling rates less than 40 kHz will invalidate the accuracy of these curves. We recommend using only 44.1 kHz or higher in order to properly realize these equalization curves.

Note 3:

Random white noise can be converted to pink noise by feeding it through the appropriate preset under the Paragraphic Equalizer.

File Conversion



The File Conversion filter is not really a filter at all but a way to convert mono files to stereo and visa-versa. It can also be used to adjust the channel balance or reverse the channels of a stereo recording, or convert a mono source into a stereo file. It is useful in converting stereo recordings made out of phase (such as old vertically recorded acoustic discs) into a stereo or mono file that is compatible with modern systems.

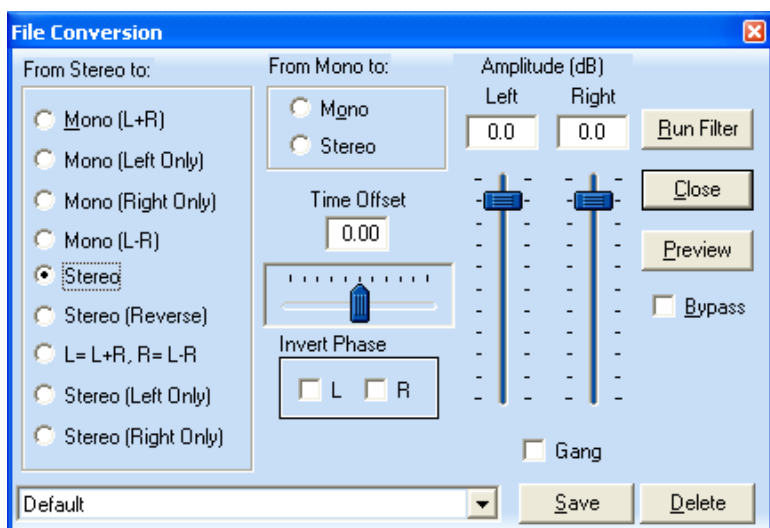


Figure 72 - The File Conversion Window

File Conversion includes two controls. One set allows you to adjust the Amplitude (gain) levels for each channel. The second control, called Time-Offset, provides a means for azimuth correction, Forensics audio enhancement, and stereo simulation. A final important use of the file conversion filter is to simply copy parts of the source file over the destination file. This is one way to revert back to the original source file (undo) following a bad filter application.

This will be the first step that you will perform in any sound restoration project. This feature provides a number of file conversion options that can provide a certain degree of noise reduction in and of themselves. Gain adjustments can also be performed during the file conversion process. The following File Conversion options are available:

From Stereo to -

- **Mono (L + R):** This option adds the .wav file Left Channel Input to its Right Channel Input before feeding it into the Destination workspace. It produces a single channel output signal.

- **Mono (Left Only) & Mono (Right Only):** These options choose only one of the two .wav file inputs to be used in the file conversion-processing step and yields a single channel output.
- **Mono (L - R):** This option subtracts the Right .wav file signal from the Left signal before feeding it into the Destination workspace as a single channel signal.
- **Stereo:** This option maintains the .wav file in stereo through the file conversion-processing step. It can be used to adjust the gains of the two signals if they are incorrect while maintaining their independence.
- **Stereo Reverse:** This option will reverse the left and right channels during the .wav file conversion-processing step. It can also be used to adjust the gains of the two signals at the same time, if they are incorrect.
- **L = L+R R = L-R:** This option provides monophonic mix to the left channel, and provides the ambient signal from a stereo recording to the right channel.
- **Stereo (Left Only):** This option takes the Left Channel input and applies it to both (stereo) output channels and yields a dual channel output signal.
- **Stereo (Right Only):** This option takes the Right Channel input and applies it to both (stereo) output channels and yields a dual channel output signal.

From Mono to -

- **Mono:** This option merely provides a clone of the original .wav file.
- **Stereo:** This option converts a monophonic single-track file into two single-track monophonic files.

Here are more details on the various File conversion options and their specific application:

- **Mono (L + R)**

This is generally used to convert a lateral cut record (like a typical 78, or a monophonic LP) which is monophonic to start with, but which has been transferred to the hard drive with a stereo cartridge, and convert these two signals into one signal on which you will perform further processing. The advantage of this simple conversion is that some of the noise content of the record will cancel out during this process, in particular, low end rumble, and even some higher frequency surface noise. This process alone can provide up to 6 dB of signal-to-noise improvement (depending on the condition of your source) compared to the use of only one of the lateral groove walls (i.e. using the left only or the right only signal).

Important Note:

It is advisable to set both gain controls to - 6 dB to avoid overloading of the Destination channel during this mixing process, unless your recording is extremely under-recorded to start with. Minus 6 dB is the default value for the two gain settings in the Mono (L + R) File Conversion feature.

- **Mono (Left Only) & Mono (Right Only)**

Sometimes, 78-rpm laterals are worn unevenly due to years of improper tracking of the tone arm that played the particular record. Therefore, it is sometimes useful to compare the Left Only groove wall with the Right Only groove wall to hear if that is the case. If you hear a significant difference between one of the two groove walls, you should then compare the quieter of the two with the Mono (L + R) signal for comparison. Choose the quietest of the three possibilities for your Destination file.

- **Mono (L - R)**

This feature takes the algebraic difference between the left channel and the right channel audio signals and feeds it into the destination file. It has three applications:

1. If you have transferred vertical cut records such as cylinders or Edison Diamond Discs utilizing a stereophonic cartridge, and haven't previously extracted the vertical signal component from that signal, this feature will enable you to do so. Just as you would have done with the laterals, it is useful to listen to

the Left Only signal and compare it with the Right Only signal to make sure that no significant tracking damage has been done to the record over the years. Choose the quieter of the two for subsequent comparison to the Mono (L - R) signal. Generally you will find that the Mono (L - R) signal has the best signal- to-noise ratio for vertical (hill and dale) recordings. Pathé (groove width modulated 78s) recordings should also be converted to monophonic utilizing the Mono (L - R) feature.

2. This feature can also be used to compensate for gain imbalances between the left channel and the right channel of the analog equipment used to make the transfers into your computer system. When listening to a lateral monophonic recording in Mono (L - R), you can adjust the gain control sliders until you hear a maximization of the noise and garble on the recording, and a minimization of the useful information content of the recording. This will provide the best setting of the gain controls when you finally make the file transfer utilizing the Mono (A + B) file conversion feature.
3. It can be used to "cancel" a television or radio broadcast out of a surveillance recording. The surveillance recording would have had to be recorded in stereo, with the surveillance signal on one track and also with a "reference" track containing the broadcast. This technique will not completely cancel out the source radio or television source, but will attenuate it somewhat. Use the gain controls and preview to obtain the most effective degree of cancellation.

- **Stereo**

This algorithm preserves a truly stereophonic Wave file in dual channel format through the sound editing and sound restoration process. It allows you to adjust the channel levels to bring them more into balance if they are not balanced to begin with.

- **Stereo Reverse**

This algorithm transposes the left and right channels from the source file before it is transferred into the destination file. This algorithm also allows you to, while reversing the channels, adjust the channel levels to bring them more into balance if they are not balanced to begin with.

- **Time Offset Feature / Azimuth Correction**

The file conversion routine includes a "Time Offset" feature. It is mounted horizontally on the file conversion control panel. For normal file conversion operations, this control **MUST** be set to zero. The time-offset algorithm provides you with the ability to retard or advance the timing between two stereo tracks. This will work with a stereo-to-stereo conversion or a mono to stereo conversion. The range of adjustment is + / - 20 milliseconds; when the control is set to its center (zero), the time offset between the two stereo channels will be zero milliseconds. To fine-tune this parameter (as with any of the parameters in the program) Use the UP ARROW key to increment and the DOWN ARROW key to decrement the time offset value. This is especially useful when performing the Analog Magnetic tape recording azimuth correction procedure.

This feature has three applications:

1. **Analog Magnetic tape recording azimuth correction:** When analog magnetic tapes are recorded or reproduced, the gap of the respective head (recording or playback) should ideally be perfectly normal (perpendicular) to the direction of the tape movement. If, in either of the two mentioned processes, the respective head gap is off-normal (off-azimuth,) two types of signal degradation will occur. The first phenomenon results in the loss of the high-end of the audio spectrum frequency response. The second effect produces a phase shift of one channel with respect to another thereby "smearing" a stereophonic image. If you are reproducing a monophonically recorded cassette tape via a stereophonic playback machine, the effect of azimuth misalignment on high frequency loss can be somewhat improved by compensating using the "Time Offset" feature (the same applies when a monophonic half-track reel-to-reel tape is reproduced on a quarter track machine.) To compensate for the effect of azimuth misalignment, adjust the "Time Offset" control until the best high frequency response is heard with your stereo system placed in monophonic playback mode while previewing. If you are dealing with stereophonic source materials, it is hard to determine the correct phasing by merely adjusting the "Time Offset" for the best image. But, if you place your

stereo system in monophonic mode with a stereo tape source, and follow the same procedure just described (use the "Time Offset" feature to adjust for the optimal high frequency response), this will also correspond to proper left channel to right channel phasing, and will therefore produce the best stereo image. Of course, you must place the system back in stereo mode after the file conversion has been completed in order to appreciate the results.

2. **Improving the Intelligibility of Forensics recordings:** It has been shown that audio signals, which are very difficult to discern, can be made more intelligible when the brain is presented with the same signal twice with a short time interval in-between. The human brain processes the information that arrives at each ear independently by the left and right brain. The information is then shared and compared between the left and right hemispheres. Comprehension of the information results from the interaction of both hemispheres communicating with one another. When a delay is injected between the information heard by the left ear and the right ear, the intelligibility factor is improved. This phenomenon was discovered by one of the British intelligence agencies (MI-5) in the late 1950's. The technique involves the use of stereophonic headphones (so that each ear is acoustically isolated from the other,) and an adjustable delay inserted between the two reproducers. The Time Offset feature can be used for this application. First, use the standard techniques for cleaning up the Forensics recording. Then, apply the monophonic signal to the file conversion algorithm using the "Time Offset" in preview mode with headsets to adjust for the best intelligibility. This may improve your ability to transcribe conversations, which would otherwise be impossible to discern. This technique does not work with loudspeaker reproduction.
3. **Stereo Simulation:** The "Time Offset" feature is one of several methods provided by DC8/DC FORENSICS that will produce a stereophonic effect. Merely start with a monophonic file, and convert it to a stereophonic file with some value of "Time Offset" applied. Adjust the "Time

Offset" control to produce the spatial effect that you desire while using preview.

- **Phase Inversion:**

180-degree phase inversion can be added to either channel by checking the appropriate box. This feature can be used to correct for out of phase stereo masters wherein the cutting head was connected incorrectly to the cutting head power amplifier. Please note that if both boxes are checked, you are right back where you started. You must only check one of the two boxes to correct for a phase-inverted channel.

- **Amplitude dB (gain):**

These two controls (one for the left channel and another for the right channel) are used to set the amplitude of each channel in terms of dB. Sometimes, these are referred to as "gain" controls. The amplitude level is shown in dB directly above each Amplitude slider control.

- **Gang:**

This checkbox will lock the Amplitude dB (gain) controls together so that they move simultaneously.

Note:

The gain and the Time Offset controls on the file converter routines can be adjusted "live" after the preview mode button has been clicked.

Cross Fade Filter



The Cross-fade filter is used to join sections of different .wav files into a single .wav file. Rather than just abruptly ending one file and starting another, the Cross-fade filter will smoothly fade from one file to another. During the time that the files overlap, the destination file is gradually faded to silence, while the source file fades from silence to full volume. This filter is also available from the Edit menu as a paste function.

Note:

This menu item and the corresponding icon will only come alive after you have files in both the Source and Destination workspace.

Using the Filter Menu Cross-fader (Tutorial)

Warning: This is not undo-able

1. Open a Source File which you desire to be the segue song in a cross-fade sequence.
2. Open a Destination File that you desire to be the first number in the cross-fade sequence.
3. Highlight the Source File up to the end of the file.
4. Highlight the Destination File segment at its ending where you desire the cross-fade to occur.
5. Click on the Cross-fade feature under the Filter Menu.
6. Set the Gain Controls as follows: (Default settings are correct)
 - File # 1 Level
 - Start = -100 dB
 - Stop = 0.00 dB
 - File #2 Level
 - Start = 0.00 dB
 - Stop = -100 dB
7. Choose the Cross-fade timing that you desire. Linear produces a pleasing result on most material.
8. Click on "Do Cross-fade."
9. The final results will reside in the Destination Workspace following processing. The entire Source File will be appended to the Destination file following the cross-fade sequence.

Virtual Phono Preamplifier (VPA™)

Phono Preamplifier Simulator including Common EQ Curves and Tone Controls

The Diamond Cut Virtual Phono Preamplifier or VPA is an IIR based digital simulation of an analog based hardware pre-amplifier having magnetic phono and line level input circuits. It uses "closed form" mathematical representations of the various EQ curves resulting in near ideal amplitude and phase response per the various EQ Standards. In addition to the standard features found on most analog circuit based

preamplifiers, it includes the ability to apply various phono equalization and re-equalization curves with near ideal accuracy. It can work in conjunction with three different types of analog input front-end (hardware) sources including standard RIAA preamplifiers, flat phono preamplifiers and line level inputs.

Besides creating the standard RIAA curve from a flat phono preamplifier front-end, the VPA's versatility allows you to use a standard RIAA hardware pre-amplifier front-end to derive from its signal the most common 78 turnover curves. It accomplishes this by reversing the RIAA curve imparted by the RIAA preamplifier front end and then re-applying the desired 78 turnover characteristic curve. It can also perform the mentioned function when utilizing a flat phono pre-amp front end*. Supported curves include the Columbia LP EQ curve found on many vintage monophonic LPs as well as two common 78 turnover curves. Decoding for acoustically mastered recordings is also provided. Additionally, a 30 Hz, third order (18 dB / Octave) Rumble filter is provided and also a mono selector switch to sum the two channels together (L+R) when transferring vintage monophonic LPs and 78s in conjunction with a stereo phono cartridge.

The VPA also includes a set of shelving tone controls (3 Band Equalizer) having a wide adjustment range so that you can tune the output of the system to your own taste if desired. And of course, it includes Volume and Balance controls.

Most soundcards produce output signal levels up to roughly 1.5 Volts RMS. And, the input sensitivity of most audio power amplifiers and powered loudspeakers require only around 1 Volt RMS to produce full output power. As a result, you can create a very simple and clean sound system consisting of nothing more than your computer coupled directly into a power amplifier and speakers or powered loudspeakers. This allows you to bypass some of the usual analog hardware preamplifiers or mixers generally found in sound-labs. This reduces the overall distortion introduced into the signal path, because the analog hardware count can be reduced which is often desirable and/or convenient. So a complete playback sound system can consist of nothing more than your computer coupled to a set of powered loudspeakers with the VPA providing a comprehensive set of controls. If the VPA is used within the Multi-Filter context in Live mode, you

can set up a complete real-time sound-system with your laptop, a set of powered speakers and an input source. Furthermore, you can also cascade other filters (like the various INF's, CNF or any of the EZ Filters) into the signal pathway after the VPA to remove impulsive and continuous noises from the source. Of course, the entire plethora of Diamond Cut filters and effects can also be added to the chain making your real-time audio system extremely versatile. It is important to note that if the VPA is used in conjunction with multiple filters and/or effects, the VPA should always be configured as the first item in the signal chain.

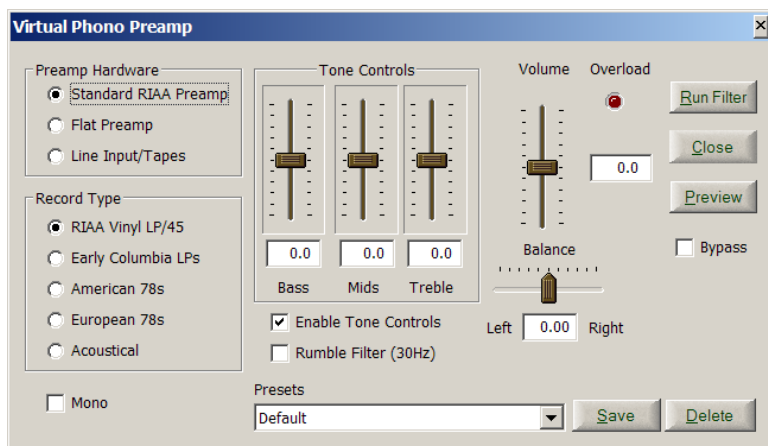


Figure 73 - The Diamond Cut Virtual Preamplifier (VPA)

Here are the VPA controls defining their functionality and range of adjustability:

Volume Control

On the top right side of the VPA, you will find a Volume control, which performs the obvious function. It has a range of adjustment from 0 to +20 dB or 0 to -20 dB and defaults to a unity gain mode (X 1.00 Gain = 0.00 dB).

Balance Control

Directly below the Volume control is the VPA channel Balance control. Normally, this would be set to the center position, which is designated as 0.00. Moving the control to the right moves the stereo soundstage towards the right hand side of your listening space, while moving the control towards the left hand side has the opposite effect. The range of adjustability for this control is 0 to -1 (when moved fully to the right) and 0 to +1 (when moved fully to the left). At either extreme setting of the balance control, the opposite channel is fully attenuated (producing no output).

Tone Controls

The VPA has three tone controls including Bass, Mids and Treble. The tone controls are enabled by placing a checkbox in the “Enable Tone Controls” selector when you desire to use them. To defeat them, simply uncheck the same selector box. The Tone Controls have the following characteristics:

Bass: Shelving type with its corner frequency set to 175 Hz and adjustability of +/- 15 dB (15 dB of Boost or Cut).

Mids: Bandpass type with its center frequency set to 900 Hz, 3 Octave Bandwidth and adjustability of +/- 15 dB (15 dB of Boost or Cut).

Treble: Shelving type with its corner frequency set to 3,000 Hz and adjustability of +/- 15 dB (15 dB of Boost or Cut).

Note concerning all controls: Fine adjustments can be made using the arrow keys on your keyboard after you have used the mouse to select a particular control.

Rumble Filter

The VPA rumble filter is of the 3rd order Butterworth variety having a corner frequency of 30 Hz. Thus, its attenuation at 15 Hz is 18 dB. It is effective for attenuating rumble sounds due to turntable (or even cutting lathe induced) record rumble, which is a low frequency random noise. Rumble is most noticeable by the random movement of your sub-woofer's speaker driver cone which wastes power and introduces inter-modulation distortion into the signal if not attenuated. It can also introduce a very annoying “wind” sound coming from a sub-woofer

system if there is too much “Rumble” present. You will know if you need this filter simply by experimenting with it. To enable the “Rumble Filter”, simply tick off the checkbox with your mouse.

Preamp Hardware (Selector Box)

The top left corner of the VPA provides you with a set of three input selections from which to choose. Your options are:

Standard RIAA Preamp: If you are using a standard magnetic RIAA phono preamplifier, then you need to check this box.

Flat Preamp: If you are using a flat preamplifier (such as the CTP - XXXX family of magnetic phono preamplifiers) you will need to check this box.

Line Input / Tape: If you are feeding your soundcard directly from a tape deck or other high-level signal source, you need to check this box. This selection will bypass the phono equalization and re-equalization systems found in the “Record Type” selector box, feeding the signal directly into the Tone Controls and / or Volume and Balance subsystem.

Please note that any of the above input hardware options need to be connected to the line input of your soundcard; do not connect the output of any of these sources to the soundcards microphone input. Line level outputs will overload the high-gain input of the microphone amplifier of your soundcard creating clipping distortion which results in a very unpleasant sound.

Record Type (Selector Box)

The lower left corner of the VPA provides a group of selections based on the type of record that you are trying to transfer to your computer. This selection, in conjunction with the “Preamp Hardware” goes through a logical truth table in order to determine the proper algorithms to run to produce the correct phono EQ curve for the particular situation that you are working with. These EQ’s and re-equalization curves are highly accurate because they are calculated in a closed form mathematical technique. The most accurate transfer results will occur when you use a flat preamplifier as your systems input front end. Analog circuit parameters associated with resistors and capacitors in

physical RIAA preamps render them less precise. These inaccuracies are reflected not only in the RIAA curve, but also in any of the re-equalization curves that the VPA can provide.

You have the following record options from which to choose:

RIAA Vinyl LP / 45

This is intended to be used on all 45 RPM records and almost all LPs which were mastered post 1955 and some that were mastered prior to that date as well. The RIAA curve was originally called “The New Orthophonic Recording Characteristic” by RCA Victor which was proposed as an EQ standard in 1953 and later adopted by the RIAA. The VPA RIAA curve is defined as having the following breakpoint frequencies:

50 Hz Pullout (3180 uSec Time Constant)

500 Hz Turnover (318 uSec Time Constant)

2120 Hz Rolloff (75 uSec Time Constant)

Early Columbia LPs

This EQ curve is intended for many early LPs mastered during the period between 1948 through 1955. This curve was established as a standard for early Columbia LPs. The VPA Early Columbia LP curve is defined as having the following breakpoint frequencies:

30 Hz Pullout (5310 uSec Time Constant)

300 Hz Turnover (531 uSec Time Constant)

1600 Hz Rolloff (99.4 uSec Time Constant)

American 78s

This EQ curve applies a turnover curve only (and the appropriate pullout) for use with 78 records having a 500 Hz characteristic. It does not implement any Rolloff, since 78s did not encode in that manner. Many American 78s were recorded with this curve. For more details regarding 78 Turnover frequencies by record brand, please refer to the “Turnover Frequency” section found in the Appendix of this user’s guide.

European 78s

This EQ curve applies a turnover curve only (and the appropriate pullout) for use with 78 records having a 250 Hz characteristic. It does not implement any Rolloff, since 78s did not encode in that manner. Many European 78s were recorded with this curve. For more details regarding 78 Turnover frequencies by record brand, please refer to the “Turnover Frequency” section found in the Appendix of this user’s guide.

Acoustical

This EQ curve reverses the curve introduced by an RIAA front-end preamplifier and is passive for signals sourced from a flat preamplifier. When used in conjunction with a conventional RIAA hardware preamplifier, this setting would often be referred to as “Reverse RIAA Mode”. It is to be used on acoustically mastered material. Most recordings mastered before 1925 require this setting, including laterally cut 78s, Edison Diamond Discs, Pathé, and cylinder recordings.

Mono Checkbox

Sometimes it is desirable to convert laterally cut recordings to monophonic (Mono L+R) to reduce surface noise pickup produced by using a stereo phono cartridge. However, often it is better to de-click your recordings first before converting your file to Mono. Nonetheless, this feature is provided in the VPA as a convenience which is especially useful when using the system as a real-time phonograph playback system. It is important to note that Hill and Dale (vertically cut) records will not play correctly using this mode. To play Hill and Dale records, please refer to the Diamond Cut File Conversion Filter (Mono L-R) to properly decode these recordings. So, when you are dealing with vertically cut records, be sure to leave the VPA Mono Checkbox turned off so that the proper decoding can be applied further down the chain of events or further down the sequence of Multi-Filter filters.

Overload

Since the VPA volume control and / or its tone controls have the ability to add gain to the system, overload (sometimes known as clipping) can occur. The red overload “LED” located in the upper right hand corner of the VPA will flash when this condition occurs. If it does, lower offending control until the Overload indicator no longer illuminates.

Presets

As with most of the Diamond Cut filters and routines, you can store your favorite settings using the preset feature in conjunction with the “Save” button. The system will prompt you to name the state of the VPA after clicking on “Save”. To recall a preset, scroll and click on the desired setting. To delete a particular preset, highlight it and then hit the “Delete” button.

You will see that there are around 60 factory presets provided in the VPA. Some presets are EQ Curves specified by their industry standard EQ Curve designation. For example, you can find the AES, NAB and FFRR LP EQ curves under those acronyms. You will also find EQ Curves that are specified by LP Label brand-name. Presets that are approximations have "Approximation" in the front of the name and finally, presets that are mathematically exact (closed form) have "Exact" in front of the name. These various LP preset curves apply to records recorded prior to the spring of 1954. Almost all LP labels switched over to the RIAA standard curve after that date. For LP records recorded later than the spring of 1954, simply use the RIAA curve found in the VPA. A complete set of VPA presets can be found in the Preset Listing section of this users guide.

VPA Application Example #1

You have an RIAA front-end preamplifier which you are using to play or transfer a European 78 RPM record. You want to decode it properly. To do so, set up the VPA in the following manner:

Preamp Hardware: Check the “Standard RIAA Preamp” setting

Record Type: Check the “European 78s” setting

Optionally, you can also tick off the Mono checkbox.

VPA Application Example #2

You are using a Flat preamplifier front-end to play Vinyl LP Stereophonic Records. To properly decode them, set up the VPA in the following manner:

Preamp Hardware: Check the “Flat Preamp” setting

Record Type: Check the “RIAA Vinyl LP/45” setting

VPA Application Example #3

You are using an RIAA front-end preamplifier to play early Columbia Vinyl LPs and you desire to decode them properly. To do so, set up the VPA in the following manner:

Preamp Hardware: Check the “Standard RIAA Preamp” setting

Record Type: Check the “Early Columbia LPs” setting

Optionally, you can also tick off the Mono checkbox.

VPA Application Example #4

You are using an RIAA front-end preamplifier to play Acoustical 78s. To properly decode them, set up the VPA in the following manner:

Preamp Hardware: Check the “Standard RIAA Preamp” setting

Record Type: Check the “Acoustical” setting

Optionally, you can also tick off the Mono checkbox.

VPA Application Example #5

You are using a Flat Preamplifier front-end to play American 78s having a 500 Hz turnover curve. To properly decode them, set up the VPA in the following manner:

Preamp Hardware: Check the “Flat Preamp” setting

Record Type: Check the “American 78s” setting

Optionally, you can also tick off the Mono checkbox.

Important Note: Never choose settings for the wrong type of Preamplifier hardware being used because the balance of the frequency

spectrum of energy distribution will be extremely unbalanced and / or distorted.

VPA Application Example #6

You are using a Flat Preamplifier front-end and you desire to play some LPs in real time through your computer sound system. You do not desire to transfer them to your hard drive, but simply to listen to them. To play them, do the following:

Bring up the Virtual Preamplifier in the Multi-Filter.

Preamp Hardware: Check the “Flat Preamp” setting

Record Type: Check the “RIAA Vinyl LP/45” setting

To play the LP in real time, click on “Live Preview” in the Multi-Filter.

Adjust the tone controls, volume and balance for the most pleasing sound.

Note: If you want to remove noise in real time while playing your LP, drag the appropriate de-noiser filters into the Multi-Filter path after (to the right side) of the VPA and adjust them appropriately.

VPA Application Example #7

You are performing a transfer using either fractional speed mastering or high speed dubbing techniques on an electrical recording and you want to decode its EQ properly. Your front-end preamplifier is of the flat variety*. To properly re-create the appropriate EQ, you will need to do the following:

Transfer the recording and then correct the speed using the Change Speed Effect. After the speed has been corrected, run the file through the VPA using the following settings:

Preamp Hardware: Flat Preamp

Record Type: Use the setting which best fits the description of the transferred recording.

*Note: The CTP Series of Flat Phono Preamps can be purchased from Diamond Cut Productions, Inc. or Tracer Technologies.

VPA Application Example #8

You are performing a 78 RPM transfer using a magnetic RIAA pre-amplifier and you are also using fractional speed mastering techniques and you want to decode its EQ properly. To re-create the appropriate playback EQ, you will need to perform the following steps in this exact sequence:

Transfer the recording to your computer and then run the file through the VPA with it set for the following:

Preamp Hardware: Standard RIAA

Record Type: Acoustical (even though the record may have been of the electrical variety; this step reverses the RIAA curve that was imparted onto the signal by your RIAA preamplifier).

Next, correct the speed of the transfer using the Change Speed Effect. Then, re-set up the VPA as follows and then “Run” the Filter:

Preamp Hardware: Flat Preamp

Record Type: Use the setting which best fits the description of the transferred 78 RPM recording.

The Effects Menu

The Effects Menu houses a good portion of the audio enhancement tools located in DC8/DC FORENSICS. This is the place to head when you want to breathe new life into an old or poorly recorded file.

Reverb



The Reverb effect is used to add a realistic room sound to a recording. The reverb is capable of simulating different size rooms, with different kinds of reflective surfaces and decay times. The reverb filter lets you

control the overall room size, decay time, early reflection level, and mix between the original material and reverb sound.

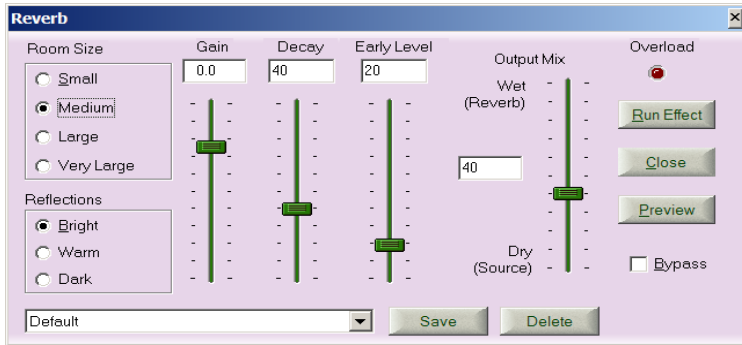


Figure 74 - The Reverb Effect Window

Reverb can be useful when dealing with recordings, which are completely “dead” as originally mastered. As with the various other filters, the reverb effect can be applied globally or selectively (using sync mode) to a .wav file. The reverb effect can also be used to convert a monophonic recording to a simulated stereophonic recording. The following controls are provided on the DC8/DC FORENSICS Reverb:

- **Room Size:** (check box)
 - Small (Club)
 - Medium (Auditorium)
 - Large (Concert Hall)
 - Very Large (Stadium)
- **Reflections:** (check box)
 - Bright: (Simulation of a very “hard” acoustical environment, as in a stone building)
 - Warm: (Simulation of a typical auditorium or theater)
 - Dark: (Simulation of a heavily draped auditorium)
- **Decay:** Control Range 1 to 99 in relative units.

The decay control affects the dampening effect of the algorithm on the reverberated signal. The higher this control is set, the longer the

reverberation “dwell-time.” The lower that this control is set, the quicker will be the decay of the reverberated waveforms.

- **Output Mix:** (Slider Control) Control Range: 0 to 100 in percentage units.

The Output mix determines the amount of the reverb effect that is fed into the system output. When the control is set to zero (dry), there will be no reverb effect. When the control is set to 99, there will only be the reverb effect, with the source signal bypassed. Useful ranges of control are usually in the 5 to 25 range, but if you are looking for extreme effects, you can get them if desired.

- **Reverb Presets:**

The Reverb is equipped with a number of descriptive presets. This is a good place to start from when using the reverb effect. Choose the desired acoustical environment (which can be selected and previewed “on-the-fly”). After you have found something close to the sound you desire, revert to the various controls to fine “tweak” the reverb for the exact sound you are looking for.

Echo Effect



The Echo Effect is a digital simulation of the olde magnetic tape delay lines (sometimes called “Tape Delay Echo Chambers”) that were popular in the early 1950’s and throughout the 1960’s. These devices worked on either of the following basic principles. One method varied the distance between the recording and the playback head producing a variable time delay, a portion of which could be fed back to the system’s input. A second method used a variable speed motor driving the tape media with the recording and playback head positions being fixed, producing similar overall results. The Echo Effect contains two independent delay lines, which can be used in a number of different modes providing it with a great deal of flexibility compared to the old analog systems. Because it performs its function utilizing digital techniques, it does not suffer from the noise and distortion buildup associated with the older analog systems. However, if the so-called “retro” sounding echo chamber is desired, you can use the Echo Effect in conjunction with the Virtual Valve amplifier in the Multi-Filter. The Echo Effect is useful in any of the following applications:

- Adding special effects to speech or music
- Creating certain acoustical simulations
- Enhancing the articulation sounds on Forensics Audio Wave files
- Simulating Stereo from Monophonic sources
- Creation of “Comb” Filters
- Adding a simple time delay to one channel of a file

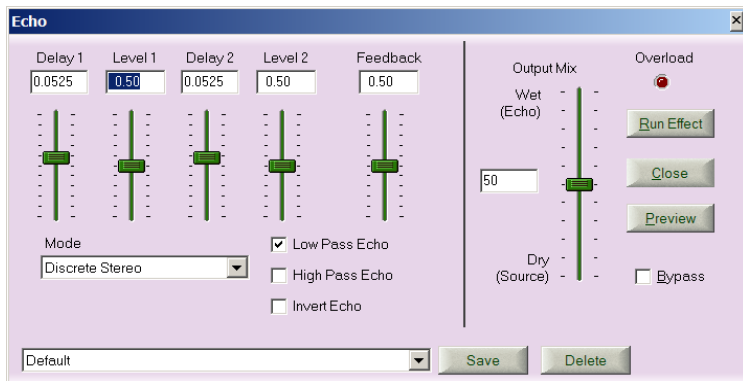


Figure 75 - DC8's Echo Effect

The Echo Effect has the following controls:

1. Delay 1: Range: 0.0001 to 5.0000 Seconds (10 uSec to 5 Seconds)
2. Level 1: This control varies the level of the effect produced by Delay 1.
3. Delay 2: Range: 0.0001 to 5.0000 Seconds (10 uSec to 5 Seconds)
4. Level 2: This control varies the level of the effect produced by Delay 2.
5. Feedback: 0.00 to 1.00 This controls the amount of signal fed from the Delay Line outputs back to its input. When this control is set to zero, you will only hear a single delay associated with each of the two delay lines. The higher that this control is set, the greater will be the reverberation sound due to feedback.

6. Output Mix: Range 0 to 100 (0 = Source Only Signal {Dry}) & (100 = Echo Only signal {Wet }). Output Mix controls the ratio of the Source signal with the processed signal. A nominal starting point for this control is 50.
7. Mode (Modes of Operation Selection Box)
 - Discrete Stereo: (Delay 1, Level 1 = Left Channel & Delay 2 Effect & Level 2 = Right Channel Effect)
 - Discrete Stereo Reverse: (Delay 1, Level 1 = Right Channel & Delay 2 Effect & Level 2 = Left Channel Effect)
 - Summed Mono – Stereo Out (Output): This mode is the same as Discrete Stereo except the input signal is summed to Monophonic before being applied to the Delay Lines. The “Dry” signal remains Stereophonic in this mode.
 - Summed Mono – Mono Out (Output): In this mode, the input signal is summed to Mono and then fed to Delay 1 Level 1. From there the signal is fed to Delay 2 Level 2 before being fed to the Output. In other words, the two delay lines are cascaded. The “Dry” signal remains Stereophonic in this mode.

Important Note: This mode can produce sustained oscillations if the feedback control is set too high.

- Overload Indicator: This indicator lights up in red when the Echo Effect system is overdriven. If this occurs, back down on the level or mix controls to prevent clipping distortion from occurring.

Ancillary Echo Effect Controls:

1. Low Pass Echo: This inserts a 7 kHz Low Pass Filter in the Effect Signal Path.
2. High Pass Echo: This inserts a 5 kHz High Pass Filter in the Effect Signal Path.
3. Invert Echo: This phase inverts the Effect Signal.
4. Bandpass Echo: This is achieved by checking both the Low Pass and High Pass Echo checkboxes. It produces a Bandpass response having corner

frequencies of 2 kHz to 8 kHz and is inserted into the Effect Signal Path.

Virtual Valve Amplifier™



The Virtual-Valve Amplifier is a computer simulation of a number of vacuum tube amplifier circuits. (Valve is the British term for electron tube. We call it the "Virtual Valve Amplifier, because that sounds cooler than "Virtual Tube Amplifier.") Its effect is to add "tube-warmth" to the sound of a recording. This is sometimes desirable to apply to DDD (purely digital) recordings. It can also be used to add subtle harmonics to very old recordings. A harmonic exciter is also included with the Virtual Valve Amplifier. It is important to note that the Virtual Valve amplifier is using real tube circuits, and real tube non-linear device characteristics to produce its effect. The wide range of adjustability of this algorithm will allow you to create an amplifier that runs the gamut in sonic performance from "grit-guitar" to "high-end audiophile."

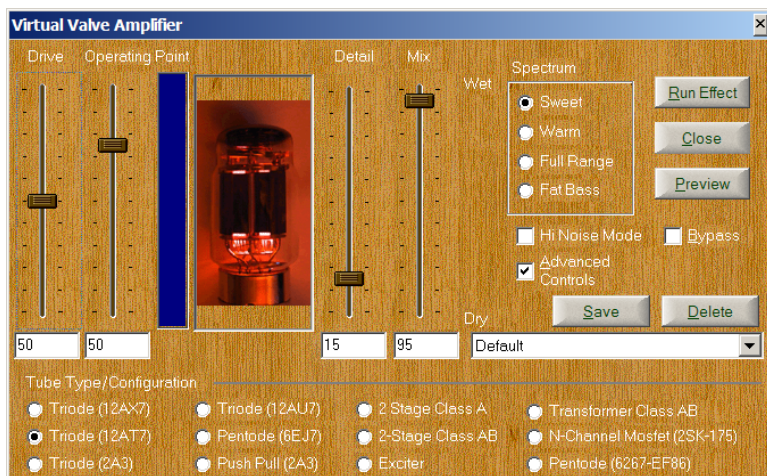


Figure 76 - The Virtual Valve Amplifier

The Virtual Valve Amplifier (VVA) produces a variety of sounds associated with valve (electron tube) based amplifiers. The effects run the range from a subtle “tube warmth” sound to extreme effects like “guitar amplifier overload” or “fuzz box.” The DC8/DC FORENSICS VVA accomplishes these effects through the use of actual electron tube circuits, which are simulated by your computer. The electronic models of the various tube amplifier circuits have been derived from the “large-signal” transfer functions of the various tubes and output transformers you can choose from. This data has been derived from extensive bench measurements of tube amplifier circuits under varying operating conditions. As such, the effects will sound literally as would be heard if you were to process a signal through a physical electron tube amplifier.

However, with the VVA, you have a great deal more control over the various sounds that can be produced, since controls, which are not normally found on electron tube equipment, have been provided. Parameters such as “Operating Point” (sometimes referred to as “Q” point by engineers) are usually fixed by the amplifier manufacturer. “Drive” is determined by how loud you play a “physical” amplifier, but with the VVA, the output level remains constant independent of drive due to an internal gain compensation algorithm. The following is a listing of the controls that are provided on the DC8/DC FORENSICS VVA:

- **Drive Slider:** 1 to 100

This control effects the degree of modulation applied to a given tube amplifier circuit and centered about the set operating point. The higher the drive level setting, the greater will be the production of predominantly even order harmonics due to the circuit’s asymmetrical non-linearity. As a result, there will be more “effect” as this control is increased. Also, the “depth” of the effect is determined in part by the degree of drive applied.

- **Operating Point (or Harmonic Control) Slider:** -100 to zero (in the middle) to +100

- **VVA Mode:**

The operating point control performs two different functions, depending on the Tube Type / Configuration selected. When a triode

or class A amplifier is chosen, it sets the operating point for the particular tube or amplifier configuration that you have chosen. Operating point also determines the device's bias value at zero signal input. The distributions of harmonics, which are introduced into the output of the amplifier, are determined to a large degree by the location of the operating point. When the control is set to + 100, (all the way up) the devices are operating close to "saturation," and when the control is set to - 100 (down), the devices are operating close to "cutoff." The non-linearity distribution is different near cutoff as compared to operation near saturation. You can use the control to achieve variations in the desired "tube effect." Most audio preamplifier tubes such as the 12AX7 are the most linear in the middle of their dynamic operating curve (control set to the middle "0" position).

- **Harmonic Exciter Mode:**

When the system is placed into harmonic "Exciter" mode, the operating point control reverts to a "Harmonics Control" which varies the distribution of harmonics that are produced by the VVA. The Harmonic Exciter is designed to provide the following audio enhancements:

- A. Synthesize the upper register harmonics that may have become lost through "generation loss" or due to the poor frequency response of the master recording.
- B. Add "presence" to a vocal recording.
- C. Create a more "up-front" sound on any modern recording.

When the control is set to +100, both even and odd harmonics are produced. When the control is set to -100, only the first 3 to 4 even harmonics of the fundamental are produced. Settings in between will produce varying combinations of the two extreme settings. The system is placed into harmonic Exciter mode by checking the "Exciter" box listed under Tube Type / Configuration, located at the bottom of the VVA window. The magnitude of the inserted Exciter effect is controlled by the "Mix" control.

- **Operating Point Indicator:**

Vertical undulations are graphically presented proportional to signal level, drive, and operating point. The Operating point indicator will have a black background in standby modes of operation and a blue

background with vertical undulations appearing in any of the operational modes of the VVA. A yellow horizontal line during operation indicates the operating point center value. Also, the dynamic operating mid-point reference is indicated by a fixed white line on the indicator. The magnitude of the drive level to the amplifier is indicated by yellow undulations plus and minus about the operation point. So both the effects of the drive and the operating point slider are indicated on the same display, for convenience.

- **Detail:** 0 to 100

The detail control allows you to control the sensitivity of the VVA to the more delicate nuances of the musical material presented and processed. The higher the setting, the greater the effect will be on the material.

- **Mix:** 1 to 100

The Mix control affects the degree of VVA signal, which is re-inserted into the signal path. At its maximum setting of 100 (wet), the dominant signal pathway is exclusively through the VVA, and when the control is set to 0 (dry), only the non-processed signal is fed through the system. You can choose any level in between which appeals to your taste.

- **Spectrum Checkbox:** -Sweet -Warm -Full Range -Fat Bass

The Spectrum Range control effects the spectral distribution of the harmonic by-products, which are passed through to the systems output. The most desirable setting is very much a function of the musical material which is being processed and the desired tube sound.

"Full Range" mode is generally used to "round out" the entire audio spectrum of a recording. It is important to note that when you are operating a tube in Full Range mode, there is a dramatic increase in the propensity for the production of intermodulation distortion, which is not particularly desirable (except for heavy metal rock). To minimize this effect, you will have to use much smaller values of "Drive" to obtain reasonable results when compared to the "Sweet" or "Warm" settings. Sweet and Warm modes are preferable to use over Full Range mode because they dramatically reduce the propensity of the electronic

valves from producing intermodulation (IM) distortion, leaving behind only the more pleasing harmonic distortion type.

Fat Bass Mode

“Fat Bass” mode is used to add some dimension to sterile sounding bass. It restricts the VVA’s action to the excitation band of frequencies below 175 Hz. However, it allows harmonics to be produced that are higher adding that “Fat Bass” sound typically associated with tube amplifier circuits. Push-pull circuit configurations will produce dominantly Odd harmonic bass effects while Class A circuits will produce both Even and Odd harmonic effects. “Sweet” and “Warm” settings restrict the operation of the VVA to the upper portions of the audio spectrum, producing pleasing tube generated harmonic and compression effects.

- **Advanced Controls Checkbox:** On/Off

This enables the more advanced controls of the VVA, if desired. If this control is not checked, default values will be chosen for some of the control settings, tube types, amplifier configuration, operating point and detail controls. Although all of those settings are preset, you will still have control over the VVA Drive and Mix settings.

- **Hi Noise Mode Checkbox:** On/Off

Some extremely noisy recordings (like certain 78 RPM records) can become even noisier when the VVA is applied to their signal. So, not only will you obtain the pleasing harmonics that you desire, but you will also obtain the undesirable elevation of the recordings already high noise floor. The “Hi Noise” mode is designed to alleviate this problem. When this checkbox is activated, the system changes into a noise suppression mode. The “Detail” control reverts into what is redefined as a “Threshold” control. Harmonics are only generated when the average harmonic signal level exceeds the set threshold which can be varied by this control. The green indicator (a virtual LED which becomes visible directly above the threshold control when operating in Hi Noise Mode) will illuminate when harmonics are being generated by the system that are above the set threshold level. Moving the threshold control upwards increases the noise reduction effect, but also reduces the effect. Moving the control all the way down will keep the VVA system active all the time with no noise suppression effect at all.

Moving the control all the way up will keep the VVA effect off all the time. Adjust the threshold control somewhere in the middle so that quiet passages squelch the system as indicated by an extinguished LED indicator. Reasonable levels of noise suppression are had when the green indicator occasionally flashes while previewing a track. This “Hi Noise” mode will reduce the incremental noise contributed by the VVA to an already noisy recording. Thus, you can enjoy harmonic generation and increased brilliance on dull & noisy recordings without suffering an elevation of their noise floors.

Please note that the Hi Noise Mode is designed to be used in conjunction with the VVA’s “Sweet” or “Warm” settings only.

- **Bypass: On/Off**

This control allows you to quickly compare the effects of the processed signal produced by the VVA to the unprocessed signal, while the program is in “Preview” mode.

- **Settings: Listing**

The VVA has a list of pre-sets, which will be a valuable starting point from which to fine tweak the adjustment controls to your desired taste. These presets are somewhat descriptive to help you in making a choice. The choices can be changed in real-time while running the program in Preview mode, so that you may compare the various presets.

- **Tube Type Checkbox:** Checkboxes for the following Valves (tubes) or circuit configurations:

- A. **Triode (12AX7)** - This configuration incorporates this high-mu dual triode into a typical RC coupled class A audio pre-amplifier configuration. This tube was chosen, because it had been and still is the industry standard pre-amplifier valve. It has a relatively flat linear operating region in the middle of its dynamic operating range, producing relatively lower levels of distortion compared to some of the other devices offered in the VVA. But, by moving the Operating Point to either the saturation or cutoff extreme, more “tube-warmth” effect can be produced by this device. This is the same device as the European type ECC83.

- B. **Triode (12AT7)** - This amplifier configuration utilizes the same type of RC coupled pre-amplifier circuit described above, but using a 12AT7 high-mu dual triode. The primary difference is that the 12AT7 was designed primarily for RF mixing applications. As a result, it has a large degree of non-linearity throughout its entire dynamic operating range, including the middle. As a result, you will be able to obtain a higher level of “evens” (even order) harmonic distortion (the most pleasing harmonic distortion) in which to add back into the signal path of the VVA. This is the same device as the European type ECC81.
- C. **Triode (12AU7)** - This amplifier configuration is simulating the driver / phase inverter stage of a push-pull power amplifier. It utilizes the 12AU7 medium-mu dual triode, and, like the previously described circuits, is biased class A and is RC coupled. This device also has a significant non-linearity in the middle of its dynamic operating curve. (In power amplifiers, some of this non-linearity is removed via the use of negative feedback, and decreasing the mix control level on the VVA simulates this phenomenon.)
- D. **Pentode (6EJ7)** - This single stage, high-gain microphone amplifier configuration utilizes a sharp-cutoff pentode. It can produce a very pleasant “tube-warmth” effect when the operating point is properly set. This device is the same as the European type EF183.
- E. **2 Stage Class A** - This is an 8 Watt class A power amplifier, consisting of a 12AU7 medium-mu triode driving a single 6L6GC beam power pentode audio output valve. Its effects are distinctive due to the convolution of the non-linearity of the triode interacting with those of the pentode, with both

devices operating in class-A mode. The 6L6GC is similar in performance to the industrial type 5881, and also the European type KT-66.

- F. **2 Stage Class AB** - This is a 25 Watt class AB power amplifier, consisting of a 12AU7 phase inverter / driver, pushing a pair of 6L6GC beam power pentodes. Because the circuit is push pull, the output devices produce a more symmetrical and reduced even-order distortion characteristic distribution. The operating point is fixed at the factory, and cannot be adjusted for this amplifier configuration.
- G. **2A3 Push-Pull** - The 2A3 is what some people refer to as a “retro – triode.” It was invented in the 1930’s, had a directly heated cathode, and produced a high power output at its time of development. It was often found used in theatrical applications and public address systems. The “Push-Pull 2A3” VVA setting uses the 2A3 triode implemented in a “push-pull” class AB₁ power amplifier circuit designed to produce 15 Watts of output power. This configuration exhibits a more linear output transfer characteristic compared to its Pentode push-pull counterpart. We have included the 2A3 tube in this particular configuration in the VVA because a musician friend of ours (Les Paul) recommended that we do so because of its unique characteristics. He explained to us that he used a push-pull pair of these devices as the power amplifier to “cut” all of the records that he released from his own home studios. The reason that he used these was the extremely clean sound that they produced. The particular devices that we used to create the 2A3 VVA models were of the “dual – plate” variety. The devices used in the characterization process for the VVA were taken from new (unused) but old stock (NOS) and were manufactured for the military by RCA Victor in 1953.

- H. **2A3 Single-Ended** - This is a single ended class A power amplifier implemented using the 2A3 power triode. It exhibits reasonably good linearity and about 4 Watts of audio in a “single-ended” class-A configuration. Its dominant distortion products are “evens.” This is the only power triode in the VVA suite of tubes.
- I. **Exciter** - This check box enables the Harmonic Exciter feature of the DC8/DC FORENSICS VVA. The exciter uses a simulation of a vacuum tube rectifier (6X4) to produce harmonics. Asymmetry between the positive and negative going transfer function establishes the relationship between the degree of even and odd harmonics produced. For more details on its performance, please refer to the Harmonic Exciter description under the Operating Point Control description.
- J. **Transformer Class AB** - This check-box enables a push-pull, transformer coupled, 6L6GC based, class AB, 20 Watt power amplifier having a 12AU7 based driver / phase inverter stage. It produces a distortion dominated by odd order components since most even order products cancel out in push-pull circuits.
- K. **6267 / EF 86 Pentode** - The 6267 / EF86 pentode was suited well for use in low-level preamplifier service where low noise and minimal microphonics were important. It was often found used as the first-stage amplifier in tape decks. Its high-gain characteristic provide it with an interesting family of operating curves that provide useful harmonic distortion and signal compression in the VVA.
- L. **2SK-175 MOSFET** - This device is not a tube, but rather it is an N-Channel Audio Power MOSFET (the P Channel compliment of which is the 2SJ-55). It was commonly found in high power, high quality

audio power amplifiers and is included here because it has a set of operating curves which differ in shape somewhat from the various VVA electron tubes. It will provide you with a different distribution of distortion harmonics which you may find pleasing in some circumstances.

Just like the other filters and effects; the VVA is equipped with a set of descriptive presets. This is always a good place to start from when using the VVA. After you have found the preset, which most closely resembles the sound you are looking to achieve, you can go back and “tweak” the controls more precisely. After you have found a group of settings that you would like to keep, use the Save Settings feature to give it a name so that you can recall it in the future.

Note: Voltage Amplification stages usually produce a phase inversion of the applied signal (180 degrees). All of the VVA devices and/or systems are phase corrected to zero degrees for user convenience.

Dynamics Processor



The dynamics processor provides you with the ability to control the dynamic signal content of the audio envelope of a .wav file. Included are compression and limiting, downward expansion/noise gating, ALC, and de-essing.

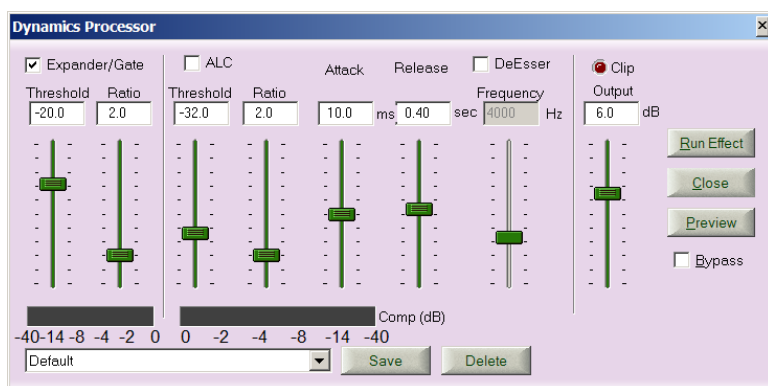


Figure 77 - The Dynamics Processor

The Dynamics Processor provides you with three functions related to the control of the dynamic range of an audio signal. The functions are as follows:

- **Expander/Gate**

(New Wider Attack & Release Time Ranges are provided in version 7 on Forward)

This system is a downward expander. When the signal is below the threshold setting, the dynamic range of the signal is increased depending on the value of the ratio setting. In other words, the incremental attenuation of the .wav file signal is proportional to the ratio setting below the threshold value. The higher one sets the value of this ratio, the greater will be the degree of downward expansion applied to the signal. Signals above the threshold value are passed through the system with no processing applied. When the ratio control is set to its maximum value (control set all the way up), the system will behave like a Noise Gate (Several noise gate presets are available in the Presets menu).

When more modest values of the ratio control are used, the system can produce some improvement in dynamic range and the average signal-to-noise ratio of a .wav file. The Expander has the following controls available:

A. Expander/Gate Checkbox: On/Off

Checking this box will enable or disable the Expander/Gate function of the Dynamics Processor.

B. Threshold: -50 dB (control down) to 0.00 dB (control up)

This control establishes the signal level below which the Expander performs its process on the .wav file signal.

C. Ratio: 1.00 (control down) to 29.99 (control up)

This control determines the degree of downward expansion applied to the .wav file for signals that are below the threshold value setting. The higher the number chosen, the greater will be the signal expansion effect.

D. Expander bar graph: Horizontal meter indicating from 0 to -40 dB.

This meter indicates the actual value of downward compression in dB, which is being applied to the .wav file signal.

E. Attack: 0.1 mSec to 1000 mSec

This control is used for the Expander/Gate, Compressor, ALC and De-Esser functions of the Dynamics Processor. It determines the time constant associated with the onset (delay) of any of the Dynamic Processor effects.

F. Release: 10 Seconds to 0.01 Seconds

This control is also used for the Expander/Gate, Compressor, ALC and De-Esser functions of the Dynamics Processor. Its setting determines the delay time associated with the decay of the particular process chosen.

- **Compressor**

This system is an upward compressor. When a .wav file signal is above the threshold setting, the dynamic range of the signal is decreased, the degree of which depends on the value of the ratio setting. In other words, the incremental attenuation of the signal is proportional to the ratio setting when it is above the threshold value. The higher that one sets the ratio value, the greater will be the degree of compression. When this value is set to its maximum, the system will behave like a Limiter. Signals below the threshold value are passed through the system with no processing applied. When the ratio control is set to its maximum value (control set all the way up), the system will produce the largest degree of compression. The Compressor has the following controls available:

A. Threshold: -50 dB (control down) to 0.00 dB (control up)

This control is similar to the threshold control for the expander, but establishes the signal level above which the compressor performs its process on the .wav file signal.

B. Ratio: 1.00 (control down) to 29.99 (control up)

This control determines the degree of compression, which is applied to the .wav file for signals that are above the threshold value setting. The higher the number that is chosen, the greater will be the effect on the signal.

C. Expander bar graph: Horizontal meter indicating from 0dB to +40 dB. This meter indicates the actual value of compression in dB that is being applied to the .wav file signal.

D. Attack: 199.9 mSec to 0.1 mSec

This control is used for the Expander/Gate, Compressor, ALC and De-Esser functions of the Dynamics Processor. It determines the time constant associated with the onset (delay) of any of the Dynamic Processor effects.

E. Release: 4.0 Seconds to 0.05 Seconds

This control is also used for the Expander/Gate, Compressor, ALC and De-Esser functions of the Dynamics Processor. Its setting determines the delay time associated with the decay of the particular process chosen.

Important Note: The Punch and Crunch Effect provides you with a somewhat different approach to compression and expansion.

De-Esser

A de-esser is a form of compressor, which is only reactive to the frequencies associated with the pronunciation of the letter “s” (“ess”). It is necessary to perform this function on over-modulated signals in the “s” frequency range that is adjustable from 1,000 Hz to 10,000 Hz. This occurs due to poor mike technique, a poor initial mix, improper mic channel equalization, or insufficient “padding” of the mic input circuit during the recording session. When the frequencies in the sensitive band are detected and are above the threshold setting, compression will be applied to the degree determined by the compressor ratio control. The attack and release controls are not active when the compressor is in “De-esser” mode. To place the compressor in the De-esser mode, click on the box by the same name.

One global control is provided in addition to all of those mentioned above. The output gain allows you to correct for overall effects (attenuation or gain) that any of the dynamic processor functions may have on the overall output signal level. Presets have also been provided to get you started with reasonable setup parameters for the various dynamic processor functions.

- **Automatic Level Control (ALC or AGC)**

The Dynamics Processor includes an automatic level control feature (ALC). Sometimes, these algorithms or systems are referred to as automatic gain controls or AGC's. This feature provides upward expansion of signals below the threshold line and downward compression of signals above the same threshold. This feature is useful in Forensics applications where there is a large variation in signal levels between several different parties that may be communicating with one another. It is also useful for the broadcast of live sporting events (if you have the DC FORENSICS version of the product) in which the crowd reaction is of interest when the announcer is not speaking. Simply clicking on the "ALC" box in the Dynamics Processor activates this feature. The threshold, attack, and release controls are still active when this function is invoked.

Output Control and Clip LED

Since the Dynamics Processor can add considerable gain to your system (especially when operating in Expander mode), an output level control and a red LED clip indicator are provided. If you observe that the red LED indicator is flashing, then the system is clipping your signal. For the cleanest results, use the Output Control to decrease the level until this no longer occurs.

Reverse File

The Reverse file feature does just that - - - it converts a .wav file so that it will play in reverse. This has several uses:

- a. Sometimes, file reversal is beneficial for removing stubborn ticks or pops with the impulse filter. By running a reversed file through the impulse filter, sometimes clicks that were otherwise too difficult to detect, may be found and removed. Of course, when the process has been completed, you must reverse the file again, so that it may be heard in the normal forward direction.
- b. The Reverse File feature may be used for reversing metal stamper recordings that have been transferred on standard turntables that were not capable of running in reverse.

Simply record the metal stamper which is reversed (using the appropriate bi-radial stylus), and then run the "Reverse File" feature to correct the transfer for forward playback.

- c. It provides an interesting effect. Of course, it can also be used to make sure that there are no demonic subliminal messages recorded in reverse on the music that your children love to listen to!

Channel Blender



The channel blender provides the ability to reduce muddy bass from vinyl recordings, decrease the "ping-pong" effect from early stereophonic recordings, minimize multi-path distortion from FM stereophonic broadcast recordings and even remove vocals from stereo recordings.

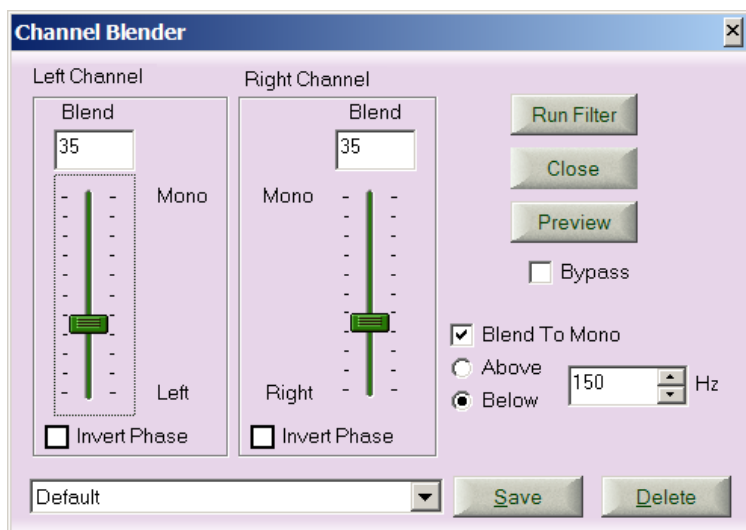


Figure 78 - The Channel Blender

The Channel Blender serves 5 purposes:

1. In the early days of stereo, channel separation was the rage. Sound engineers often literally segregated the recording artists into separate recording studios in order to maximize the channel separation. This later became known as the “ping-pong” effect. In other words, with stereo separation, if a little was good, and more was better, then too much was thought to be just enough. The Channel Blender can be used to reduce the extreme stereo separation found on some of these early stereophonic recordings, restoring them to a more natural sound.
2. Rumble on Vinyl recordings is dominated by the vertical displacement component of the master recording and playback stylus. Since bass is acoustically non-left or right below about a hundred Hertz, this rumble can be reduced by summing the signals and then adding them back into the main stereophonic channels below a certain crossover frequency. The Blend to Mono feature performs this function when it and the “below” function are checked. This can add clarity and improved bass definition to vinyl recordings, which sound muddy due to excessive rumble. Keep in mind that rumble is not just a by-product of the turntable from which you are playing a record, but also involves the system which mastered it in the first place. Even though you may have a very expensive turntable, you will still encounter recordings that are laden with rumble. The recommended frequency for this feature is around 125 Hertz with the “below” box checked. Experiment to determine the best results for the material that you are dealing with.
3. FM stereo multi-path distortion, when it occurs, is dominant in the last two octaves of the audio spectrum. By placing the channel blender in Blend to Mono above the corner frequency setting, you can reduce this distortion with a tradeoff of channel separation at the upper end of the audio spectrum. Try corner frequency settings starting at around 5 kHz with the “above” box checked.
4. Lastly, ambience can be enhanced on a stereo recording by phase inverting one of the channels and summing by

the L-R rather than the L+R information back into the main signal path. This is accomplished by phase inverting one of the two channels.

5. It can reduce the “thump” that you hear when you play a cracked record by blending both channels to 100.

The Channel Blender has the following unique controls:

- **Left and Right Channel Blend Controls:** These two controls take the summed or differenced signal and add it back into the respective left and / or right channels. When these controls are set to 0, there is no blending effect. At a setting of 100, the blending is maximized. An Invert Phase check box is located in the Left and Right blend control panels. This produces a 180-degree phase inversion of either of the channels before the summation takes place. Therefore, you can blend in L+R (with the phase inversion boxes not checked) or you can blend in the L-R signal (with ONE of the two Invert Phase boxes checked). The L-R signal contains the ambience information on most stereophonic recordings. If both Invert Phase boxes are checked, the signal reverts back to L+R, so if ambience enhancement is desired, only check one box.
- **Blend to Mono Checkbox:** This checkbox sums the signal to monophonic above or below the indicated frequency. You can select a crossover frequency anywhere between 10 and 10,000 Hertz.
- **Above:** This blends to mono all frequencies above the corner frequency setting. This is used to reduce multi-path distortion from FM broadcasts.
- **Below:** This blends to mono all frequencies below the corner frequency setting. This is used to reduce rumble and muddy bass on vinyl stereophonic recordings.

Punch and Crunch™



Punch & Crunch is a multi-channel dynamic compressor, expander or ALC. It allows you to modify the dynamics of a piece of audio in a very natural sounding manner. In some cases, you may want to restore lost transient response by applying the expander mode, but in other cases you may want to compress the signal in order to produce more “broadcast presence.”

Punch and Crunch is a four band dynamic expander (Punch) and compressor (Crunch) and automatic level control (ALC). It is useful for a number of applications such as the following:

1. Adding dynamic range or “Punch” back into severely compressed radio broadcast or vinyl recordings.
2. Adding “dial presence” or “Crunch” (compression) to radio broadcasts, without suffering the “pumping” effect found in conventional wide-band dynamic compressors.
3. Decreasing the dynamic range of classical music so that it can be more “listen-able” in restaurant or automotive environments by applying the compressor function.
4. Improving the signal-to-noise ratio and dynamic range of old 78-RPM recordings by using Expander mode.
5. Improving the intelligibility of forensics recordings.
6. Special Effects creation.



Figure 79 - Punch and Crunch in Fixed Frequency Mode

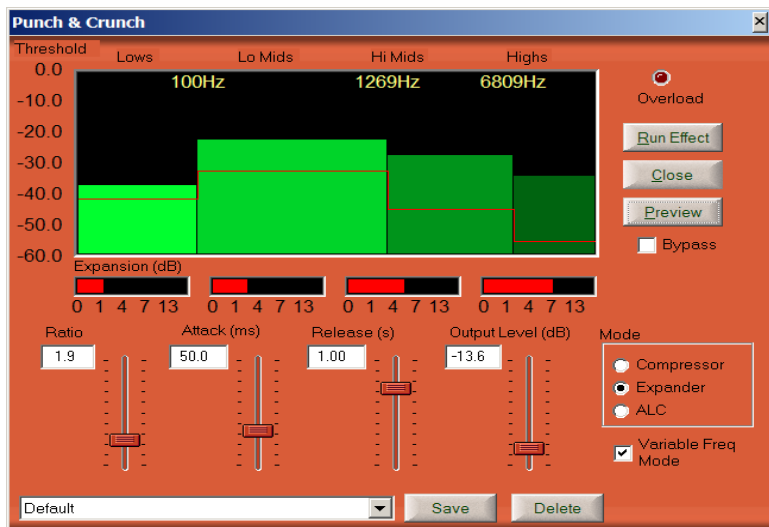


Figure 80 - Punch and Crunch in Variable Frequency Mode

It works by breaking the audio spectrum into four separate bands. Each band is independently expanded or compressed when its signal exceeds the graphical display of its particular threshold line. Using the ratio control modifies the degree to which the bands are expanded or compressed. The actual compression or expansion of any particular band is shown by horizontal bar graphs for each band that are calibrated in dB. The bands are broken into the following “buckets”:

- **Band 1:** 0 to 125 Hz
- **Band 2:** 125 Hz to 900 Hz
- **Band 3:** 900 Hz to 4,000 Hz
- **Band 4:** 4,000 Hz to 20,000 Hz

The following controls are provided on Punch and Crunch:

- **Graphical Display of the Four Bands.** Each band is represented on this graph and the incoming signal present on each band will modulate the vertical displacement of each band. Threshold for each band can be dragged with the left mouse button to the desired position. When the threshold is dragged all the way to the top, that band will have no dynamic compression or expansion effect. When a band is dragged all the way to the bottom, it will have a maximum compression or expansion effect.
- **Graphical Display of the Expansion or Compression of each band.** This graph is horizontally modulated and located beneath each of the four bands. It is calibrated in dB. It will tell you the amount of compression or expansion being applied to its associated band.
- **Ratio:** This controls the degree of compression or expansion applied by the system. When the system is operating in compression mode, you can choose up to 30:1 compression. When the system is operating in expansion mode, you can choose up to 15:1 expansion.
- **Attack:** This determines the time constant associated with the onset of this effect and is calibrated in mSec.

- **Release:** This determines the time constant associated with the decay time for this effect and is calibrated in Seconds.
- **Output Level:** This allows you to adjust the output level of the system. Use this in association with the Overload indicator to minimize clipping distortion.
- **Mode:** This allows you to choose either Expansion (Punch) or Compression (Crunch) modes of operation. Additionally, you can choose ALC (Automatic Level Control) mode, which is often useful in Forensics Audio applications. It can be used to improve the intelligibility of speech having wide variations in signal level resultant from multiple persons speaking simultaneously on a recording.

Variable Frequency Mode

This mode allows you to shift the location of all of the crossover frequencies anywhere across the audio spectrum. You are not tied into the fixed frequency associated with the legacy mode of operation for the Punch and Crunch effect. Because the frequency ranges are user definable, they can be placed quite close together and so the system also employs symmetrical third order slopes at the various crossover frequencies, unlike the asymmetrical nature of the fixed frequency mode of operation crossovers. This provides better isolation between bands thus reducing adjacent band crosstalk. To place the system into the variable frequency mode of operation, merely tick off the “Variable Frequency Mode” checkbox. Then, you can drag from side to side the various crossover frequencies (the vertical lines associated with the bar graphs). And, just like in fixed frequency mode, you can vary the threshold at which each band becomes active by dragging the horizontal component of the bar graphs upwards or downwards.

Change Speed



The Change Speed Filter is designed for use in several applications:

- It can be used to correct the speed of a recording.
- It can be used for fractional speed mastering. This feature is important in any of the following three instances:

1. You have available a 45 RPM turntable but do not own a 78 or an 80 RPM turntable.
 2. You want to play 80 RPM records, but only have a 78.2 RPM turntable without variable speed.
 3. Your turntable will play all speeds, but the record you are attempting to transfer is so warped that the stylus skips off the record due to the vertical undulations.
- It can make speech transcription easier by slowing it down to the rate at which you can scribe. Also refer to the Stretch & Squish filter for this functionality. Note: The Change Speed tool will alter the pitch of the audio, while the Stretch and Squish Tool will lengthen or shorten an audio file without changing the pitch.
 - It can be used to produce interesting special effects.
 - It can be used to “tweak” a musical piece to meet a contests time requirement for a choreographed performance.

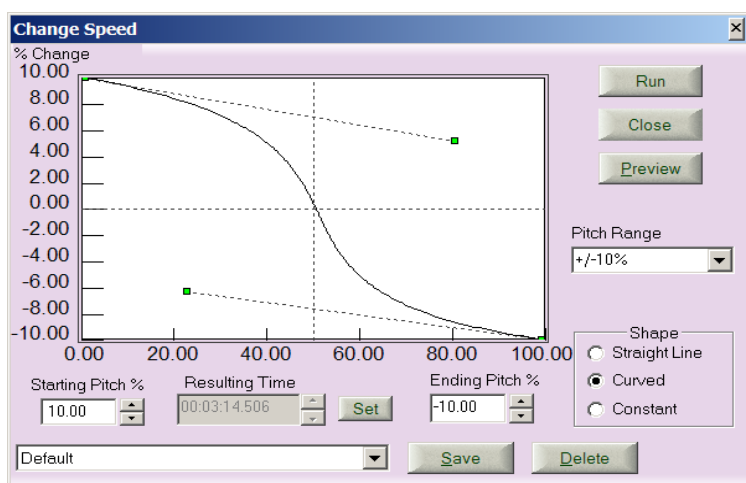


Figure 81 - Change Speed

The following is a summary of the control parameters and the range of adjustment provided for the Change Speed filter:

- **Starting Pitch Control:** -50% to +100%
- **Ending Pitch Control:** -50% to +100%
- **Display Pitch Range (3 ranges):**

1. +/-1%
 2. +/-10%
 3. +100 / -50%
- **Shape (Pitch vs. Time):**
 - Straight Line (2 Green Cursors)
 - Curved Line (4 Green Cursors)
 - Constant (Start and End Track each other)
 - **Graphical linear or curvilinear Pitch inflection points**
(green square cursors on graph)

The Graph shows you how you have programmed the speed to change as a function of the selected .wav file time axis. You can use the mouse to drag the two green cursors to establish the time relationship that you desire. Often, a flat line is appropriate; however, sometimes the speed of the cutting lathe would slow down towards the end of the recording. The reason this occurred is that some of the early recording lathes used wind up mechanical motors with governors rather than electrical hysteretic synchronous motors. To correct this defect, a pitch decrease (negative pitch slope) is necessary towards the end of the recording. When the curve shape is selected, two additional green cursors appear. The new green cursors can be moved both vertically and horizontally allowing you to create numerous curvilinear pitch vs. time relationships. Curvilinear correction is useful for fixing the speed of non-capstan based tape recordings that have been transferred using a capstan-based machine. It can also be used to create interesting special effects when used in conjunction with the looped preview or looped play mode.

Use the following Formulae to calculate the % Pitch Change required when using fractional speed mastering:

Wherein:

T = Actual Turntable Speed (RPM)

&

R = Rated Recommended Record RPM (see RPM Chart in the Glossary of Terms section of this manual)

then:

$$\Delta \text{ Pitch (\%)} = ((R/T) - 1)(100)$$

One method for determining the pitch change required for a particular recording is to measure the line frequency “Hum” on the signal. This can be accomplished with your Diamond Cut Spectrum Analyzer. Deviations from the ideal values (50 or 60 Hz) can be compensated using the Change Speed feature by applying simple ratio proportions based on the measured hum frequency value compared to the ideal value.

Note: The Change Speed Effect will not appear in the Effects list if no file has been opened.

Automatic Change Speed Compensator

Sometimes, it is hard to determine what the correct pitch correction factor that should be used for a particular recording when the line frequency hum of the original recording can’t be discerned from other noise components. But, if you know the length of the piece, you can use the Automatic Change Speed Compensator to correct its pitch. Often, the time duration of a particular track is published in the liner notes associated with the recording. Simply do the following:

A. Make sure that the recording is trimmed carefully with no lead-in or lead out dead-time. You can always add back into the file some silence after the speed compensation process has been completed.

B. Click on “Set” and then enter the published value of time into the “Resulting Time” data entry field.

C. Run the filter and thereafter the pitch should have been corrected so long as the published time duration for the track is accurate.

This method can also be used to take a track that may be a little too short or too long for a choreographed performance and correct it per the contest rules (within reason). Another feature to consider for this type of situation would be the Diamond Cut Stretch and Squish time compression algorithm, especially if large time changes are required.

Note: The Automatic Change Speed Compensator should only be used in Straight Line Slope mode.

Time Compression And Expansion (Stretch and Squish)

The Stretch and Squish filter is a variation on the theme of the Change Speed filter. It allows you to change the cadence (tempo or speed) of a piece of audio while maintaining its pitch at a constant value. You can either slow down the beat or cadence of a piece (Stretch) or speed it up (Squish).

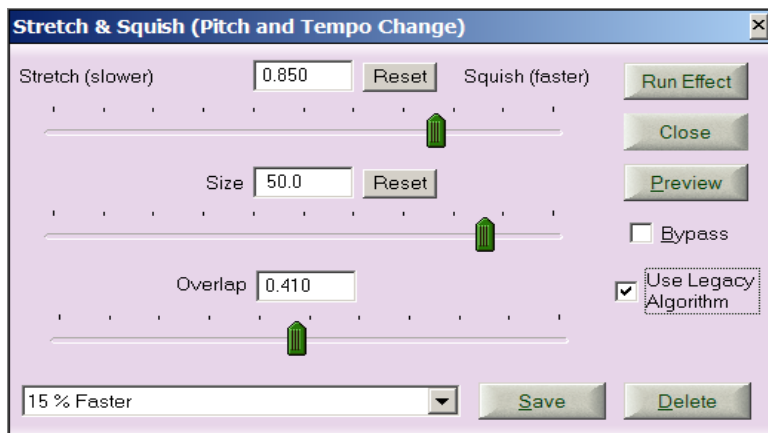


Figure 82 - Stretch and Squish shown in Legacy Mode

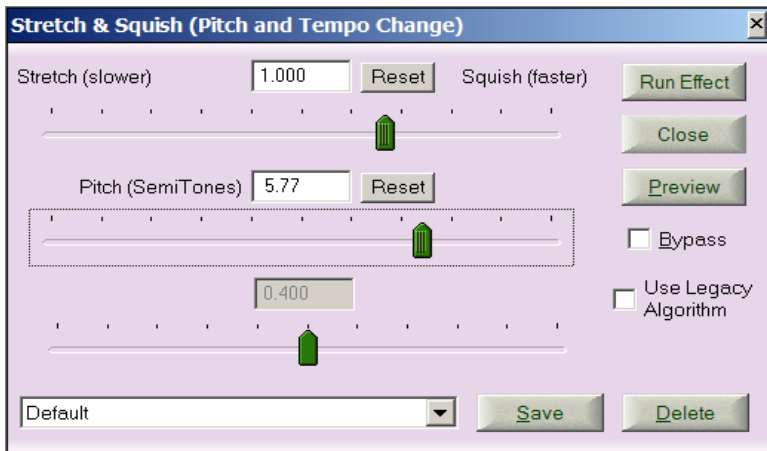


Figure 83 - Stretch and Squish shown in Pitch Shift Mode

Stretch & Squish is actually a special case of the speed change filter. It has the ability to maintain a relatively constant pitch while modifying the cadence or beat interval of the .wav file that is being processed. Its primary purpose is for Forensics applications in which a spoken word recording needs to be slowed down for transcription to the written word; however, it can also be used to learn a fast guitar riff by slowing it down, or for other special effect situations. It has a range of adjustability that will allow you to half the cadence in “Stretch” mode which means that it is slowed down. Also, it has the ability to double the cadence of a .wav file in “Squish” mode. Stretching or Squishing a .wav file is accomplished with the Stretch control. If you set the Stretch control to 1.000, the file will play or preview at normal speed. However, if you decrease the number to its lower limit of 0.5000, you can double the speed of the recording. If you increase the number to its upper limit of 2.000, it will now play at half speed. Also, a Size control is provided which allows you to vary the update rate of the algorithm. This control affects the overall sound quality of the filter, which is subjective in nature and left up to the user. The following controls are provided on the Stretch & Squish Filter:

- **Ratio:**
0.500 to 1.000 speeds up the file and represents “Squish” mode

1.000 to 2.000 slows down the file and represents “Stretch” mode

- **Size Control:**

20-240 (*mSec*)

- a. 20-100 (Only useful for special effects)
- b. 100-240 (Useful for Speech and Music without significant distortion)

- **Overlap Control:** 0.2 to 0.6 (Determines the normalized degree of frame overlap and must be adjusted qualitatively. The best sound will typically be found between .3 and .5)

Pitch Shift Mode (Semi Tones): (Range - 12 to + 12) In Pitch Mode (operable only when Legacy Mode is unchecked) you can change the pitch of a file while keeping the tempo (rhythm or cadence) constant. Its range of adjustment is +/- 1 octave wherein +12 represents a one octave increase in Pitch and - 12 represents a one octave decrease. The calibration is in semitones meaning that the octave above and below the source material is divided in twelfths (12 semitones in an octave). Each unit of measurement associated with the Pitch control represents one note on the chromatic musical scale. You can fine-tune the pitch of the system using the left and right arrow keys in association with the Pitch control. To do so, just click on the pitch control slider and then use the up and arrow keys until you achieve the desired pitch while previewing your file. The exact semitone value being used is displayed in the digital readout located directly above the Pitch control.

The Stretch & Squish Effect includes a set of 6 “Disguised Voice Effect” presets that you can use to shroud human voices behind a sonic veil (making them unrecognizable). You should not “hot switch” between these presets while previewing them, because some of them actually change the algorithm being used to accomplish its effect. Always stop the preview process before selecting another one of these presets to assure proper algorithm initialization.

Note:

This filter produces digital artifacts. Artifacts are normal for this type of mathematical manipulation when applied to a .wav file. The greater

the amount of “Stretch” or “Squish” applied, the greater will be the degree of artifact generated by the routine. Quality will decrease as you increase the amount of compression or expansion.

In “Pitch” mode (operable only when Legacy Mode is unchecked) you can change the pitch of a file while keeping the rhythm or cadence constant. Its range of adjustment is +/- 1 Octave wherein +12 represents an octave increase in Pitch and - 12 represents a 1 Octave decrease. The calibration is in semitones meaning that the Octave above and below is divided in twelfths (12 semitones in an Octave).

Filter Sweeper



The Filter Sweeper is a novel system providing a variable frequency response versus time capability. The sweeper can be applied to an entire .wav file or any portion by using our selective filtering capability. Three filter types are available in the Filter Sweeper including a sweepable Low-Pass, High-Pass, and Notch. Besides producing interesting special effects, the Filter Sweeper has a number of very useful audio restoration applications. Here are two examples.

The hum frequency of a recording sometimes changes as a function of time. This audio defect may be due to the fact that the original analog tape recorder was battery powered and as the battery became weaker during the recording process, its speed decreased. This mechanism will have caused the resultant hum frequency to increase versus time when played back later on a constant speed system. To correct this defect, you can apply the Filter Sweeper set for operation in “Notch” mode. Set the “Start Frequency” to the appropriate lowest hum frequency value of the defective .wav file, and the set the “Stop Frequency” to the highest value of hum frequency. These two values of frequency can be determined by using the Spectrum Analyzer while using the “looped play” mode during the appropriate sections (beginning and end) of the defective .wav file. The “Sweep Type” will need to be determined empirically, because the appropriate compensation curve will depend on the type of capstan servo motor regulator that was used in the offending tape recorder in conjunction with the type of batteries that had been employed.

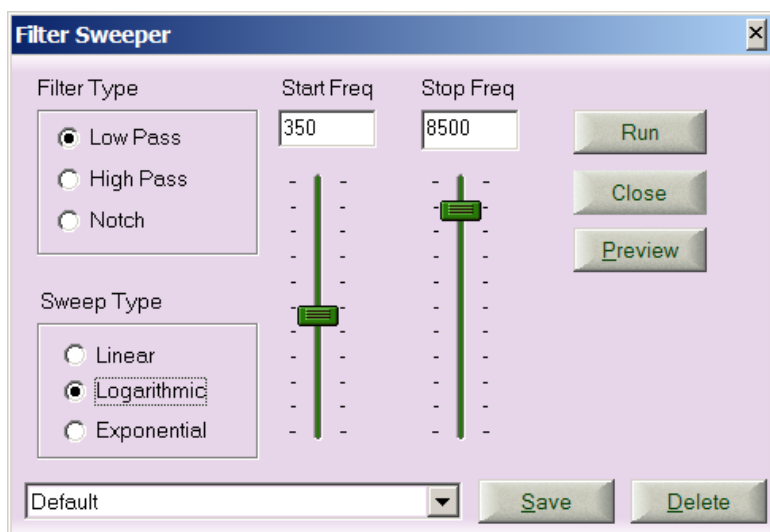


Figure 84 - The Filter Sweeper

Another application of the Filter Sweeper is what we call “masked” Fade-Ins and/or Fade-Outs. These “masking” techniques not only can be used to produce very interesting effects, but also provide a very useful noise reduction tool. How can a swept filter produce noise reduction? The principle is based on a simple musical concept coupled with the non-linear nature of the human sense of hearing. Some musical material commences with very simple progressions (including only one or two musical instruments), then moving into a series of complex riffs and refrains (maybe including a few crescendos) and then often diminishing in complexity towards its ending (soft curtain). Much of the noise in such a recording is hidden by the complexity of the musical material found in the central body of the piece. The noise on the recording is heard to be most prevalent to the human ear at its beginning and its ending during the compositions simple parts. (In actual fact, the noise level is probably relatively constant throughout the entire recording.) That is why we refer to this phenomenon as “masking. This perception is a trick that the human ear plays on the listener which you can take advantage of as an audio restoration engineer. By first selecting an area at the beginning or end of the piece and then using the various Filter Sweeper presets labeled “Masked Fade-In” or “Masked Fade-Out”, you can smoothly contour the

response of the restoration system during these quiet intervals at the beginning and ending of the musical rendition. If there is excessive hiss present during these time periods, use the Low-Pass Sweeping filter. If there is a high degree of low frequency noise (like rumble and or HVAC noise), use the High-Pass Sweeping filter. If there is a combination of both of these types of noises, you will have to apply two passes of the Swept filter to obtain the optimal results, running one operation with the Low-Pass and another with the High-Pass or by using the Multi-filter. Since some musical material is simple at its beginning and end, reduced frequency response will often not be observed by the listener due to the application of this new technique. The “Sweep Type” that you choose will depend on the type of musical progression that you are dealing with, and is also somewhat subjective.

The Filter Sweeper provides the following controls:

- **Filter Type:**
 1. Low Pass (First Order Butterworth)
 2. High Pass (First Order Butterworth)
 3. Notch (Second Order with 0.1 Octave Bandwidth)
- **Sweep Type:** (determines the shape of the sweepers frequency versus time relationship)
 1. Linear
 2. Logarithmic
 3. Exponential
- **Start Frequency:** (Determines the frequency value in Hz from which the sweep begins.)
Range: 10 – 19,999 Hz
- **Stop Frequency:** (Determines the frequency value in Hz at which the sweep terminates over the determined highlighted area of the .wav file.)
Range: 10 – 19,999 Hz

Sub-harmonic Synthesizer



Sub-harmonics are fractional multiples of a fundamental frequency. Some audio media have difficulty recording the lowest octaves of the audio spectrum and thus are deficient in this sonic area. The Diamond Cut Sub-harmonic Synthesizer is designed to help correct this audio anomaly. It can also be used to add “deep bass” to any recording. The sub-harmonic synthesizer takes audio signals in the bass portion of the spectrum and divides those frequencies in half and then adds them back into the systems output mix.

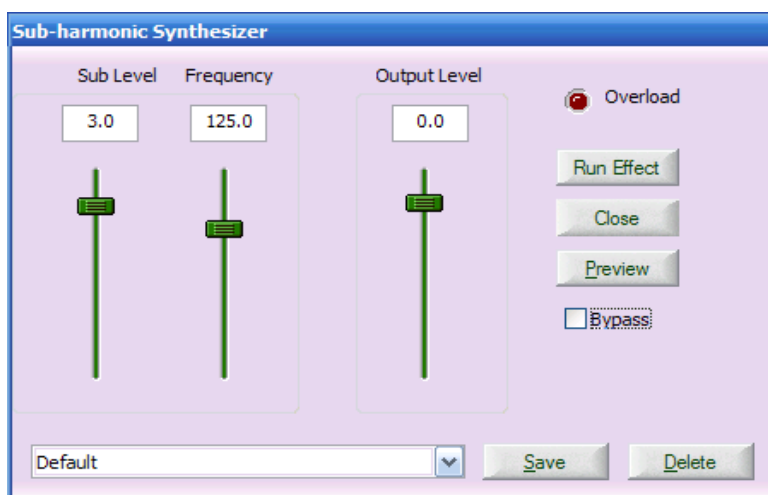


Figure 85 - The Sub-harmonic Synthesizer

You can choose the upper frequency limit at which this process takes place via the “Frequency” control. A good frequency to begin with is 125 Hz; adjust to your personal taste using the Preview button. You can also adjust the wet/dry mixture of the synthesized sub-harmonics with the original source signal by way of the “Level” control. As an example, if the sub-harmonic synthesizer frequency control is set to 125 Hz, frequencies are generated in the range of 62.5 Hz on down to the bottom end of the audio spectrum. These signals are then added back into the main signal pathway thereby re-constituting bass (or deep bass) notes which may be missing from a recording.

The Sub-harmonic Synthesizer includes the following user controls:

- Frequency: Range is 60 Hz to 150 Hz – This controls those frequencies below which are used to excite sub-harmonics. It is the “corner frequency” setting.
- Sub Level: Range is -60 dB to + 15 dB – This controls the amount of the effect applied to the output of the effect.
- Output Level: Range is – 60 dB to +10 dB – This controls the overall output level of the effect which includes both the wet and dry signals. Keep this control set so that the “Clip” LED does not illuminate.

Note 1: For “Ultra Deep” bass effects, you can cascade two sub-harmonic synthesizers together in the Multifilter. Set the second one in the chain to half the frequency value of the first. For example, set the first one to 150 Hz and the second one to 75 Hz.

Note 2: Sub-harmonic distortion can be introduced by this effect on male vocals, especially when the frequency control is set to values greater than 100 Hz.

Overtone Synthesizer



Overtone Synthesizer are multiples of a fundamental frequency. Some audio media have difficulty recording the upper octaves of the audio spectrum and thus are deficient in this sonic area. The Overtone Synthesizer effect is designed to help correct this type of audio anomaly. The Diamond Cut Overtone Synthesizer operates in the upper portion of the audio spectrums and is the compliment to the Sub-harmonic synthesizer. It can be used to add treble or “brilliance” to any recording. It can also be used to enhance the hissy – sibilant sounds sometimes missing from highly muffled forensics recordings. The Overtone Synthesizer takes audio signals in the treble portion of the spectrum and multiplies those frequencies times two and then adds them back into the systems output mix.

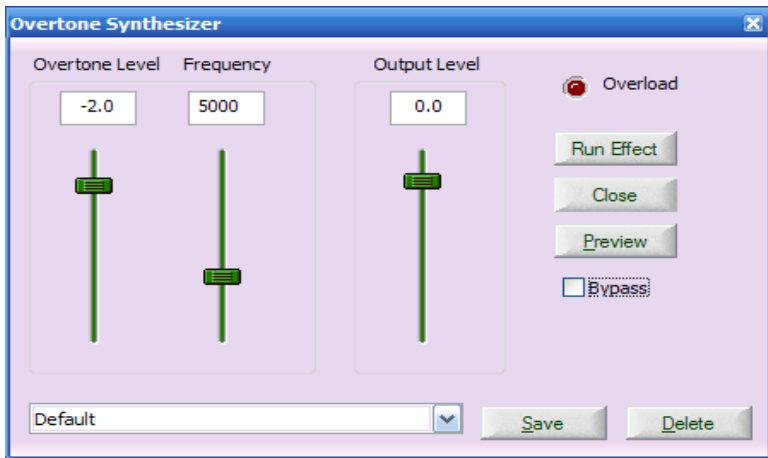


Figure 86 - The Overtone Synthesizer

You can choose the lower frequency limit at which this process takes place via the “Frequency” control. A good frequency to start with is 5,000 Hz; adjust to your personal taste. You can also adjust the wet/dry mixture of the synthesized sub-harmonics with the original source signal by way of the “Level” control. As an example, if the Overtone Synthesizer frequency control is set to 4,500 Hz, frequencies are generated in the range of 9,000 Hz on up to the top end of the audio spectrum. These signals are then added back into the main signal pathway thereby re-constituting treble (or brilliance) which may be missing from a recording.

The Overtone Synthesizer includes the following user controls:

- Frequency: Range is 3,500 Hz to 9,000 Hz – This sets those frequencies above which are used to excite overtone generation. It is the “corner frequency” setting.

Note: The range for this control in the Forensics version is from 2,000 Hz to 9,000 Hz.

- Overtone Level: Range is -60 dB to + 10 dB – This sets the amount of the effect applied to the output.

- **Output Level:** Range is – 60 dB to +10 dB – This sets the overall output level of the effect, including both the basic signal as well as the overtones. Keep this control set such that the “Clip” LED does not illuminate.

Note 1: For “Ultra Brilliant” treble effects, you can cascade two Overtone Synthesizers together in the Multifilter. Set the second one in the chain to half the frequency value of the first. For example, set the first one to 4,000 Hz and the second one to 8,000 Hz.

Note 2: Muffled Forensics recordings can be enhanced via the “Forensics Pseudo Sibilant (clarifier)” preset. Adjust the “Overtone Level” for the optimal result. Also, please refer to Note 3.

Note 3: The Overtone Synthesizer is limited for use on files that were sampled at 22.05 kHz and above. This fact is especially important in Forensics Audio applications which use lower sampling rates. If you desire to use it on files that were mastered on files that were recorded below 22.05 kHz, up-sample them to 44.1 kHz first and then apply the Overtone Synthesizer.

EZ Enhancer™



The EZ Enhancer provides an easy-to-use method for adding some spice to your recordings. It combines three types of audio enhancement algorithms into one system including harmonic generation, dynamic processing (compression or expansion) and tonal contouring. A large number of factory presets are provided which produce combinations of various harmonic generation types coupled with dynamics signal processing. The names of these various presets are descriptive and so should make choosing the correct one for a given job quite intuitive. The presets work in conjunction with the “Exciter” and the input “Level” controls. Three tone controls are also provided including Bass, Mids and Treble. The system is capable of remembering the state of the EZ Enhancer system controls based on your own preferences via the “Save” feature. Here is a description of the EZ Enhancers controls:

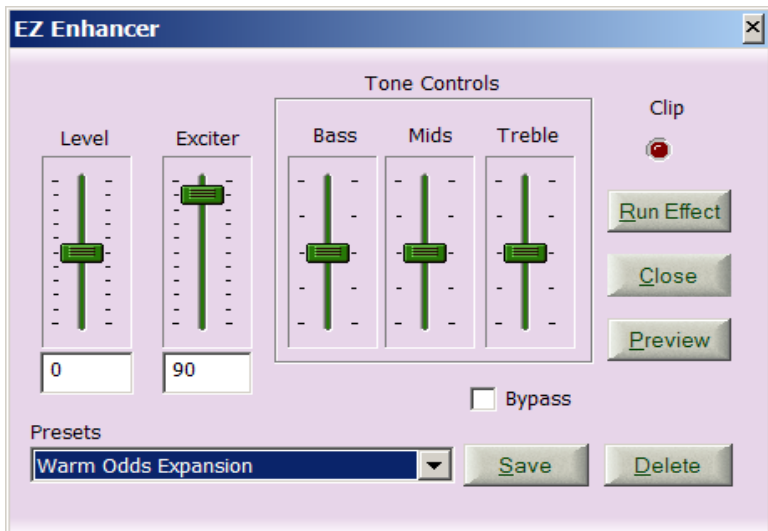


Figure 87 - The EZ Enhancer

Level Control: -100 to 0 to +100 wherein 0 represents no gain added on the input side of the dynamics processor sub-system. Since the dynamics processing elements of the various presets are non-linear in nature, you will find that there is an interaction between the Exciter and the Level Controls in terms of sound tonality. Adjust both for the most pleasing result. The level control does not directly affect the overall level of the system, but only as a higher order effect. 0 is the default setting for this control and in most cases will produce satisfactory results without any “tweaking”.

Exciter Control: This control affects the degree to which the harmonics generator and the dynamics processor output signals are applied to the source signal. The range is 0 to 100 wherein 0 represents no enhancement (dry) and 100 represents the maximum enhancement (wet).

Bass Control: This is a shelving type of control having its corner frequency fixed at 175 Hz and having an amplitude adjustability range of +/- 15 dB (15 dB of boost or cut).

Mids Control: This is a Bandpass type of filter having its center frequency fixed at 900 Hz and having a bandwidth set for 3 octaves. It can produce up to +/- 15 dB (15 dB of boost or cut).

Treble Control: Like the Bass Control, this is also of the shelving variety having its corner frequency internally set to 3,000 Hz and having an amplitude adjustability range of +/- 15 dB (15 dB of boost or cut).

Clip Indicator: A red clip indicator is provided to show if the dynamic range of the EZ Enhancer system is being exceeded. If this indicator illuminates, then you must find which control is set too high and lower it until the indicator extinguishes. Otherwise, clipping type distortion will occur, which is not specifically very pleasant sounding.

Note: The best results will be realized by applying the “Normalize Gain Scaling” feature (found under the CD Prep Menu) to – 6 dB before running the EZ Enhancer.

The Forensics Menu

Surveillance, two-way radio, and noisy telephone communications place special burdens on noise reduction software. DC8 and DC FORENSICS Audio Laboratory comes complete with a special set of filters aimed at Forensics applications that can be found under the Forensics Menu. Besides being useful in Forensics applications, these filters can be used for a variety of non Forensics-related material. The DC Forensics Audio Laboratory version has a more extensive and higher performance feature set in this regard compared to its DC8 counterpart.

Noise encountered in Forensics applications can be categorized into several basic categories.

- **Out-of-band noise:**

This type of noise is rejected with the brick-wall band-pass filter.

- **In-band repetitive noise at a level lower than the target signal:**

This type of noise is rejected with the brick-wall band-stop filter.

- **In-band random noise at a level lower than the target signal:**

The standard Continuous Noise filter found under the Filter menu rejects this type of noise.

- **In-band repetitive noise at a level equal to the target signal:**

The Adaptive filter set up for the Keep Residue mode of operation can attenuate this type of noise. If this noise is in the form of a “buzz” you should also consider trying the Harmonic Reject filter found under the Filter menu.

- **In-band random noise at a level equal to the target signal:**

This type of noise can be reduced by using the Adaptive filter set up for the Normal (random) mode of operation or the DSS.

Several of the Forensics filters can also be used in those extreme circumstances such as when the noise level is significantly greater than the “good” signal level. In this instance, the ability to hear and understand the underlying speech is typically the goal – not a complete restoration. See the Note below.

Note 1:

It is important to note that the Forensics filters are not optimized specifically for “high fidelity” applications, but more for improving the intelligibility of speech or the discernment of subtle sounds buried in noise.

Note 2:

The Diamond Cut Forensics Filters perform best on files that are 44.1 kHz or higher. If your native file is encoded with a lower sampling rate, up-convert it first before using the Diamond Cut Forensics Filters. While this will not improve the bandwidth of the signal, it will speed up filter processing time.

EZ Forensics Filter™ (Forensics Version Only)



The EZ Forensics Filter is a very good place to start when working with any Forensics audio file. It is very easy to use and often will provide a good solution to your noise problem, even if not 100 percent optimal. It combines a dual question wizard scheme coupled with a high degree of automation, adaptive filtering techniques and 170 pre-programmed filter solutions. The EZ Filter process will help you to hone-in on the best solution to a Forensics Audio problem. It will present you with a few filters to choose from after you answer a few questions. By previewing your file and comparing the alternative filters presented, you can then choose the best filter for the job at hand.

A graphical display shows you the amplitude vs. frequency of the input signal to the filter. The black line shows the user modified output response on the output side of the EZ Forensics filter. The adjustable blue inflection point (square dot) is the user interface for adjusting the output response of the system after the best filter has been selected.

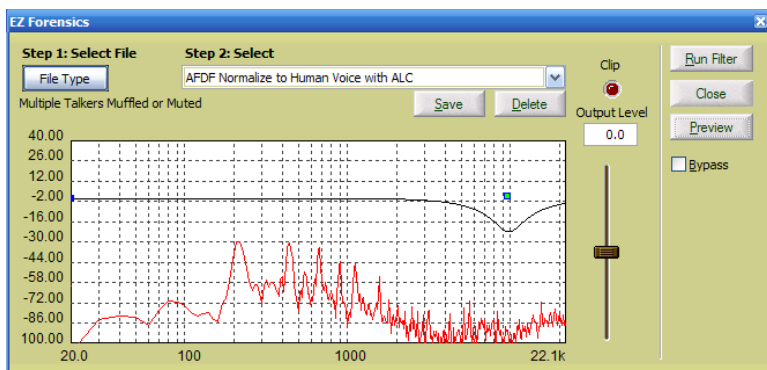


Figure 88 - The EZ Forensics Filter

The basic process for using the EZ Forensics Filter is as follows:

1. Answer the First Question found in “Step 1: Select File Type”. Click on the “File Type” button and a “Recording Type” Dialog Box will appear. Choose (by left mouse clicking) between the alternatives based on the nature of your audio source:

- Single Talker
- Multiple Talkers
- Telephone or 2-Way Radio Communications
- Background Sounds
- Binaural (2 Mic) Decoding

2. Click on “Next” in the “Recording Type” Dialog Box and then answer the second question presented in the “Recording Problems” Dialog Box. You can choose between:

- Something Else Not Listed Here
- Hissy, Sibilant High Frequency Noise
- Low Frequency Rumbling Noise
- Muffled or Muted Sound
- Power Line Hum
- Buzz
- Steady Tones
- Variable or Swept Frequency Tones

3. Under “Step 2:” you will notice that a number of filters are presented to you. Choose the best one by “Previewing” each one and decide on which one does the best job.

4. You can “tweak” the response of the system by left-mousing the blue inflection point on the graphic display to provide emphasis or attenuation anywhere on the frequency spectrum that you desire. Moving the mouse horizontally changes its frequency while moving the mouse vertically amplifies or attenuates the signal at that frequency. You can also adjust the “Output Level” with the slider control having the same name. Be sure that you do not “clip” the output to avoid signal distortion. A clipping condition is indicated by the illumination of the red “Clip” virtual LED indicator.

5. If and when you are satisfied with the results, then highlight the file or a portion thereof and click on the “Run Filter” button.

Now, Wasn’t That Easy?

A description of the filter type being used is announced just below “File Type” button. If the EZ Forensics Filter did not provide the optimal result, the DC Forensics Audio Laboratory provides you with a wide array of alternative filters that you can customize for a specific Forensics Audio application. They can be used individually (one at a time) or in concert with one another via the Diamond Cut Multifilter.

Advanced EZ Forensics Filters

To “Go Advanced” with the EZ Forensics approach, a good place to start is with the following two Multifilter Presets:

EZ Forensics_Protol (A Time Domain Adaptive Filter Based Approach)

EZ Forensics_Protol2 (An Adaptive Frequency Domain Filter based Approach)

These two Multifilter presets provide you with the basic signal path structure used in the EZ Forensics system. But, in the Multifilter, you have complete control over a myriad of parameters which you can customize and tweak to your specific needs. Consider using this approach in the event that the EZ Forensics Filter is not providing the optimal results that you seek.

Note 1: It is important to convert all forensics .wav files (8 kHz, 11.025 kHz and 22.05 kHz) up to a 44.1 kHz sampling rate before using the EZ Forensics Filter. To accomplish this, use the Change Sample Rate feature found under the Edit menu.

Note 2: The best results will be realized by applying the “Normalize Gain Scaling” feature (found under the CD Prep Menu) to – 6 dB before running the EZ Forensics Filter.

The Adaptive Filter (Forensics Version Only)



This Time Domain Adaptive Filter (TDAF) is used largely in audio applications where the ambient noise environment is constantly changing and the filter co-efficients must automatically adapt to

maintain good intelligibility of an audio signal. This is the Time Domain based “older sister” to the forensics Adaptive Frequency Domain Filter (Forensics AFDF) found in the Diamond Cut Continuous Noise Filter.

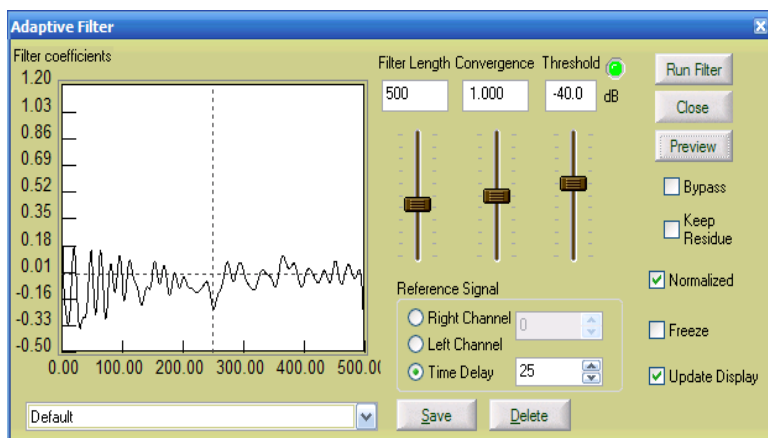


Figure 89– The Time Domain Adaptive Filter

The Adaptive filter adjusts itself to remove a modeled signal representing the unwanted time domain waveform while preserving the target signal. It uses an advanced form of an adaptive least mean squared algorithm to provide continuous adaption of the filter coefficients. It works best with a reference signal (in other words, a stereo or binaural source) containing only the noise to be rejected. This second channel or reference signal can be obtained from a second surveillance track with its microphone located near the noise source in the room such as a Jukebox or a television set. However, it can also use its own signal as a reference in conjunction with the time delay function, which is provided for monophonic situations. Additionally, the adaptive filter provides either the main processed signal or a keep-residue mode output signal for rejecting a wide variety of different types of noise sources. Sometimes it will be found that the “keep residue” mode signal is more useful than the main output signal. Trial and error is sometimes the best way to determine the best mode to use. The following controls are included with the adaptive filter:

- **Convergence(Adaptation Speed)** - Slower adaptation speeds produce better noise rejection for stationary noise (not changing), while faster adaptation speeds produce better adaptation response in quickly varying ambient noise conditions.
- **Filter Length (Samples)** - The larger this number the more signal inflection points can be modeled in the time domain signal in order to be rejected or maintained. Generally speaking, a “sweet spot” is often found between 10 and 100 samples. Very long filter samples can be very time consuming of processor resources.
- **Reference Signal** - This is the signal to be compared to when a stereo recording is available. (Choose the one that contains the reference signal to be used.)
 1. Right Channel
 2. Left Channel
 3. Time Delay
- **Time Delay (Samples)** - For use in time delay reference mode only when no reference signal is available. This essentially allows a delayed representation of the signal being adapted to be its own reference.
- **Adapt / Freeze button** - Selects adapt or freeze coefficients mode of operation. Usually this is activated in Adapt mode for the first 5 seconds of noise. In some cases where the sound ambient is constantly changing, one may choose not to “freeze” this filter.
- **Keep Residue button** - Allows the operator to use the error signal rather than the output signal from the Adaptive filter. This feature is useful for attenuating continuous loud or varying tones that may be masking a Forensics recording.
- **Threshold** - This control sets the level above which the system re-initializes itself based on the applied signal amplitude. Generally, you will find that settings

somewhere between -20 dB and -40 dB are useful values. If this control is set to 0 dB, it will not produce any output because it will be continuously re-initializing itself.

- **Threshold LED Indicator** - Just to the right of the threshold control is a Green LED indicator. When this indicator is illuminated, the adaptive filter is active and adapting to the signal that is present. If the LED is not illuminated, move the Threshold Control downwards until the Green LED flashes or illuminates in order for proper filter operation to occur.
- **Graphical Display** - You can choose between two graphical display modes. One mode displays the Amplitude of the Filter Co-efficients on the vertical axis (on a normalized basis) vs. the Index of the filter Co-efficients (which is an indicator of how many filter taps or filter length which are being used). The second mode plots the frequency response (relative amplitude vs. frequency) being produced by the adaptive filter on a time-updated basis.
- **Adaptive Normalize** - This button, when checked, places the system into a “Normalized Least Mean Squared” mode, which is useful when attempting to clean up material with widely varying signal amplitude components. This is the preferred default mode of operation.

Important Note:

Because of the wide range of convergence allowed in the Adaptive Filter, certain audio signals may cause the filter to become unstable and cut out. In these cases, try changing the Convergence parameter until the audio is restored. Usually lower convergence is more stable, but adapts more slowly to changes in the audio. Lengthening the filter may also increase stability.

Brick Wall Filter



These filters differ from their “Hi-Fi” counterparts in that they are very steep digital filters used to attenuate signals that are interfering with a poor quality audio signal. These are FIR (Finite Impulse Response) filters that exhibit a very high degree of out of band attenuation. These filters are predominantly used for forensics audio applications.

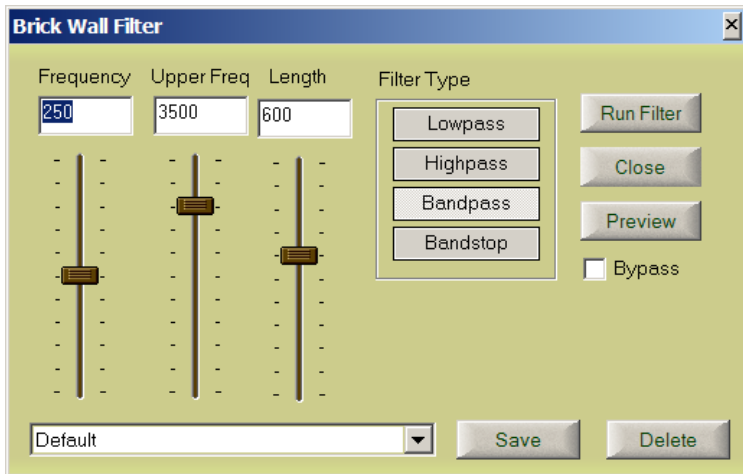


Figure 90 - The Brick Wall Filter

The Brick Wall Filters include the following choices of filter shapes:

1. **Lowpass:** Only allows signals below the corner frequency to be fed to its output.
2. **Highpass:** Only allows signals above its corner frequency to be passed to its output.
3. **Bandpass:** Allows only the signals between its chosen lower frequency and its upper frequency setting to be passed to its output.
4. **Bandstop:** Rejects all signals between its selected Frequency and Upper Frequency limits.

Several controls are provided:

- **Frequency** (Hertz) (Range is from 3 Hz to 20,000 Hz)
- **Length** (Samples up to 4094) - The larger the value of this parameter, the greater the degree of rejection past the filter's corner frequency setting. However, the higher the setting for length, the larger will be the computational demands on your computer. So, large values of "length" will take longer to process or may make your machine "stutter" in preview or live real time modes.
- **Upper Frequency Control**- Sets the upper limit for the filter's bandwidth. (Range is from 3 Hz to 20,000 Hz)*
- **Lower Frequency Control**- Sets the lower limit for the filter's bandwidth. (Range is from 3 Hz to 20,000 Hz)*

You can monitor the signal being removed by the brick wall filter by previewing its reciprocal filter. In other words, if you are using the Bandpass filter, switch the system to the Bandstop filter to hear what is being removed. Similarly, if you are using the Lowpass filter at a certain frequency, switch to the Highpass filter at the same frequency to hear what is being removed. The same logic applies to the Highpass filter; just use the Lowpass filter set for the same frequency to hear what is being removed.

***Note** - Only applies to the Bandpass and the Bandstop filter.

Polynomial Filter (Forensics Version Only)



This allows mathematicians, scientists, and engineers to create their own transfer function using a polynomial expression. For those not so inclined, there is a plentiful assortment of presets to choose from. Also, there are two methods of data entry possible, either numerical or graphical.

This system realizes a transfer function in terms of the input to output signal ratios. It is useful for canceling the non-linearity's that may have been introduced during a recording process. Essentially, this feature can be useful in some circumstances for reducing harmonic distortion that was created by the recording process non-linearity. The method of data entry for the Transfer Function is in the form of the coefficients associated with a 5th order polynomial for which the actual transfer

function graph is plotted automatically. You can enter the coefficients numerically for each term or you can use one of the numerous presets that are provided to get you started with a particular function. It is useful to have some mathematical background in order to effectively use this filter using co-efficient data entry. However, trial and error is the best method for finding a setting that will reduce the distortion of a particular recording since the recording non-linearity is not generally known ahead of time. Furthermore, sometimes adding non-linearity's can be useful to enhance the intelligibility of extremely muffled conversations. Limited instantaneous dynamic expansion and compression can also be realized by using this feature, for which there are several presets provided for convenience. Interesting distortion creation and frequency multiplication is also possible with this algorithm.

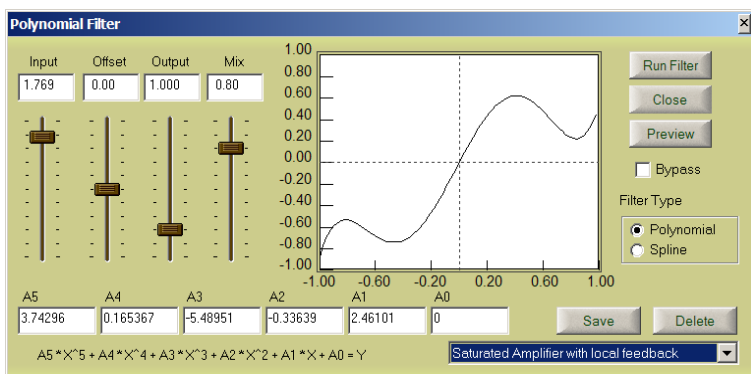


Figure 91 - The Polynomial Filter

1. The Transfer Function is given in the general form by the following polynomial expression:

$$Y = A5X^5 + A4X^4 + A3X^3 + A2X^2 + A1X + A0$$

You have a field in which you can enter each of the co-efficients of the polynomial, including:

$$A5 = \underline{\hspace{2cm}} \quad A4 = \underline{\hspace{2cm}} \quad A3 = \underline{\hspace{2cm}} \quad A2 = \underline{\hspace{2cm}} \\ A1 = \underline{\hspace{2cm}} \quad A0 = \underline{\hspace{2cm}}$$

Note: Values of A can be positive or negative.

- **Input Gain:**
Range: 0.001 to 1.999 (1.000 represents unity gain)
This controls the level of the signal being applied to the input of the polynomial expression.
- **DC Offset :**
Range: -1.00 to 0 to +1.00 (0 represents no offset value)
This control effects the DC offset applied to the input signal to the equation.
- **Output Gain:**
Range: 0.001 to 5.000 (1.000 represents unity gain)
This control affects the output of this system after the polynomial equation has been applied to the signal.
- **Mix:**
Range 0.000 to 1.000
This control affects the degree to which the processed signal is added to the input signal for presentation to the systems output. 0 represents no polynomial effect and 1.000 represents complete polynomial effect on the output.
- **Filter Type:**
You can choose between “Polynomial” or “Spline” modes via the radio buttons. In “Polynomial” mode, you create your transfer function by entering your data as co-efficients ranging from **A0** to **A5** in the appropriate numerical fields. If you choose “Spline”, you can create the transfer function you desire graphically by manipulating (with your mouse) the 4 square shaped inflection points presented on the graphical display.

Spectral Filter (Forensics Version Only)



This filter is essentially a very high resolution Graphic Equalizer that uses FFT techniques. It includes four EQ modes of operation from which to choose. In manual mode, it allows you to create a very high-resolution frequency response contour containing up to 32,000 bands of equalization (essentially a 32,000 Band Graphic Equalizer). The user interface system is intuitive and allows you to zoom-in on a particular portion of the spectrum that needs accurate and specific frequency response contouring. By using the right mouse button, you can add bands or inflection points, or delete them simply by pointing and clicking on the graph. This is very useful in Forensic audio applications for removing in-band and out-of-band extraneous noises because of its high degree of frequency selectivity and its very steep slope characteristic.

The spectral filter also includes a spectral inverse filter mode which can be used either manually or automatically. This feature measures the signal amplitude per frequency bin in a .wav file and then applies an inverse response curve in order to normalize it to a reference contoured shape. Essentially, it reverse normalizes to constant signal amplitude per unit bin against a user selected (by preset selection) curve, or via a user customized curve. In other words, frequency bins with lower signal levels than the reference curve are amplified until those bands equal the reference and those frequency bins with greater amplitude than the reference curve are reduced in amplitude until they match the reference curve. One could view this as an automatic equalizer or an Auto EQ. Spectral inverse mode is very effective for creating an automatic equalization of poorly recorded forensics audio files. This can result in dramatically improved intelligibility of distorted or muffled sound files. It can be viewed as a speech clarifier. File sampling for this feature can be performed manually by highlighting the area of interest or automatically by way of the use of the Auto Sample (checkbox) feature. In Auto Sample mode, the spectral inverse filter becomes essentially an adaptive equalizer and continuously re-samples your .wav file (or signal in the case of Live Preview mode) on the fly. Live Preview mode is accomplished via the Multifilter feature in your Diamond Cut suite of filters.

The Spectral Copy mode of operation of the Spectral Filter allows a .wav file to be normalized in terms of frequency content per unit bin to another file or to a certain section of a .wav file. To use the Spectral Filter in this manner, simply highlight a portion of a .wav file while in Spectral Copy mode and then click on the Sample Spectrum button. Next, bring up another .wav file and either Preview or Run the filter. The spectral response of one file will then become imposed on the second file. This mode may be found to be useful when an exemplar of a particular sound exists in one file, but is suspected to be buried in the noise of a second file. An examiner could then sample on the exemplar file containing only the sound of interest (its sound-print) and use that response to help amplify signals in that area of the spectrum on a second file in order to make certain signals more discernable or intelligible.

A fourth Spectral Filter mode of operation is called “Spectral Difference” (sometimes referred to as Spectral Matching or EQ Matching). In this mode, you can impose the frequency response of one file onto another. This is useful if you want two independently recorded sound tracks to sound similar in terms of frequency response. Simply put, you can cause one .wav file to sound sonically similar to another. To use this mode, you must perform the following operations after switching the Spectral Filter EQ mode to “Spectral Difference”. You will note when this operation is performed, that a new button appears labeled “Sample Source”. Before proceeding, normalize the gain of both .wav files to the same value (this feature is found under the CD Prep Menu).

1. Bring up your reference .wav file (the file which you want your other file to sound like). This first file is your “Source” file which should be the sonically higher quality of the two files that you will be working with.
2. Highlight a sector (around 5 to 30 seconds) of this source reference file. The sample should be taken somewhere in the middle of the file. It should not be taken at the noisy lead-in sector of the file. Better signal averaging results from longer sampling time values.
3. Click on the “Sample Source” button and the system will

then perform some calculations culminating in the creation of a green colored graph which represents the frequency response of the sample.

4. Next, bring up your target .wav file (the file requiring frequency response modification).

5. Click on the Spectral Filter Preview Button and you will notice that the system will draw two more graphs.

6. You will then hear the corrected response of the Target .wav file.

7. Optionally, you can highlight an area of the Target .wav file and press the “Sample Spectrum” button. This will take a sample of the target file and compute a difference spectrum.

8. The frequency response of the Target .wav file is shown in Blue on the graphic display.

9. The red spectral curve is the difference spectrum and is also the equalization curve that will be applied to the target wavefile.

Upper and lower crossover frequencies are settable in the “Spectral Difference” mode of operation by way of the “twin green goalposts” which are visible on the graphical display. These vertical green lines can be “dragged” horizontally along the frequency axis using the left mouse button. The left-most “goalpost” controls the lower crossover frequency while the right-most “goalpost” controls the upper crossover frequency. The frequencies in-between the “goalposts” are the values for which the spectral difference is calculated and is reflected by the shape of the red graph. Those signals above and below the “goalpost” crossover frequencies revert to the native target file values. Adjust these goalposts until you achieve the most natural sound with the least amount of high frequency hiss and low frequency rumble.

Another mode of operation which may be found to be useful in some applications (having widely varying target frequency responses) is the

Auto Sample button option. This will update the calculation of the spectral difference on-the-fly providing you with a variable spectral difference compensation system.

Note 1: Forensics Audio Work may benefit from high values of FFT size while studio audio work (or “Hi-Fi” work) may benefit from substantially lower values of FFT size.

Note 2: The Spectral Difference filter can be used in a variety of ways. You can use just one file and sample different portions to generate a difference spectrum or use two different files and apply the difference to a third file.

Note 3: When using Spectral Copy and/or Spectral Difference modes of operation, both .wav files must have the same sample rate and bit depth attributes.

Note 4: When using the Spectral Difference mode to match the response of a noisy file to a clean file (Cassette and CD for example). It is important to do all of the restoration work on the noisy file before applying the spectral difference filter otherwise the noise components will be detected as signal and distort the desired response.

Note 5: When using Spectral Copy and/or Spectral Difference modes of operation, both .wav files must use the same FFT Size setting.

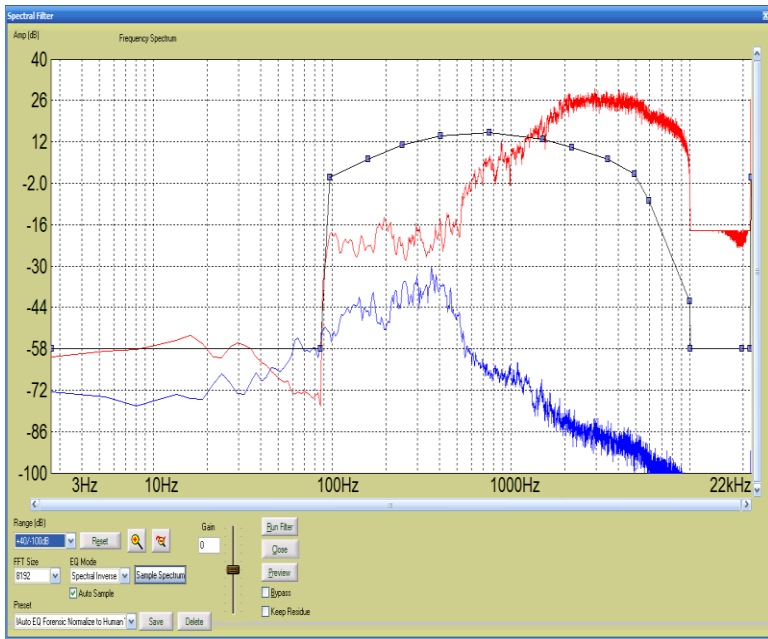


Figure 92 - The Spectral Filter

The following controls are provided in the Spectral Filter:

- **Amplitude vs. Frequency Graph:** Up to three graphical lines are shown on the Spectral Filter graph. In Manual mode, a black line with square inflection points (or touch points) are shown. In Spectral Inverse Filter mode, the blue line represents the frequency response of the wavefile (or a portion thereof). The Red line represents the calculated Inverse Spectrum response required to correct the response to the black line curve.
- **Graphic Representation of the user defined Frequency Response:** Drag the blue inflection points (dots) with your mouse in order to establish the desired frequency response.

- **Range (dB):** This parameter determines the vertical axis range for the Spectral Filter. The following ranges are provided:
 1. +40 / -100 dB
 2. +20 / - 40 dB
 3. + / - 40 dB
 4. + / - 20 dB
 5. + / - 10 dB
 6. + / - 3 dB

- **FFT Size(number of Frequency Bands):**Higher Values of FFT Size improves frequency resolution / frequency discrimination but with the tradeoff of higher levels of digital artifacting. Choose the best balance between those two requirements based on the requirements of the file you are working on. The following FFT size selections are available in the Spectral Filter:
 1. 256 (128 Bands)
 2. 512 (256 Bands)
 3. 1,024 (512 Bands)
 4. 2,048 (1,024 Bands)
 5. 4,096 (2,048 Bands)
 6. 8,192 (4,096 Bands)
 7. 16,384 (8,192 Bands)
 8. 32,768 (16,384 Bands)
 9. 65,635 (32,768 Bands)

- **Zoom In:** Click on the Magnifying Glass Icon with the “+” sign or drag the mouse over the area of the spectral graph on which you desire to focus.

- **Zoom Out:** Click on the Magnifying Glass Icon with the - sign.

- **EQ Mode:** Allows you to choose between the four main modes of the spectral filter. They are Manual Adjustment, Spectral Copy, Spectral Inverse and Spectral Difference. Manual Adjust is a direct manipulation of the frequency response curve similar to a standard Equalizer. Spectral Copy creates a response curve that is the shape of the average response of the sampled file. Spectral Inverse creates an EQ response that seeks to normalize the response of a

signal to a desired frequency response curve (or contour). Spectral Difference creates an EQ response that represents the difference in frequency response between two samples.

- **Auto Sample Checkbox:** This feature enables an adaptive mode of operation in which the system automatically samples and applies a portion of the .wav file to the Spectral Inverse or Spectral Difference Filter for normalization / correction on an ongoing basis. This feature is particularly useful in “moving mic” situations when the acoustical environment is changing throughout the recording. This mode of operation will help optimize the intelligibility of the wavefile in these variable acoustical environment situations. You can freeze the sampling any time desired by un-checking the checkbox while previewing the file. If you freeze the Auto Sample feature, the last sound-print will be held in memory thereafter and will be used as the reference response when the filter is “Run”.
- **Gain Control Slider:** Adjusts the Spectral filter overall output level over the range of -40 dB to + 40 dB.
- **Bypass Check box:** Allows you to hear the source material with the Spectral Filter out of the system.
- **Keep Residue:** Allows you to hear what you are removing when using the Spectral Filter. Mathematically, the keep residue signal is the Input Source Signal minus the Spectral Filtered Signal.
- **Reset:** Returns all of the blue inflection points (bands) on the graphical display to a “flat” (white noise related) response.
- **Sample Spectrum Button:** This control is used in conjunction with the Spectral Inverse Mode of operation. It allows you to take a sound-print (manually) of a highlighted sector of your wavefile for use as the signal to be used to create the desired spectral inverse filter response.
- **Sample Source Button:** This button is used to sample a source reference file’s frequency response and is only appears when the system is operating in Spectral Difference Mode.

- **Presets:** (Save or Delete buttons) - The factory presets are especially useful when using the Spectral Inverse Filter Mode in that Normalization curves are provided to White, Pink, Inverse Pink, Brown and Inverse Brown noise contours with ease. Other useful curves are also provided. For definitions of the mentioned noise types, please refer to the Glossary of Terms section of this Users Guide.
- **Right Mouse Button:** To use this feature, select the function you desire with the right mouse button and move the band with the left mouse button. Here are the functions available for the Spectral Filter by using the Right Mouse Button:
 1. Add Point (adds a frequency band on the graph where you are pointing with the mouse)*
 2. Delete Point (deletes a frequency band on the graph where you are pointing with the mouse).
 3. Reset Point Count (to the system default value - - - factory default = 0 dB, flat-line response)

*Note 1: Adding frequency bands can also be accomplished by double-clicking the left mouse button.

Note 2: It is important to convert all forensics .wav files (8 kHz, 11.025 kHz and 22.05 kHz) up to a 44.1 kHz sampling rate before using the Spectral Filter. To accomplish this, use the Change Sample Rate feature found under the Edit menu.

Spectral Filter Application Example #1

You have a Forensics Audio file that is extremely muffled, meaning that its intelligibility is extremely poor because of a lack of the sibilant sounds due to significant high frequency loss. To improve the intelligibility of this file, place the Spectral Filter into Spectral Inverse Mode and check the Auto Sample checkbox. Choose the Auto EQ, Normalize to White Noise or the Auto EQ, Normalize to Human Voice preset. Preview the file. If the sound is improved to your satisfaction, then Run the Filter. If the improvement is not adequate, experiment with some of the other Auto EQ based factory presets until you find the best one for your purposes.

Spectral Filter Application Example #2

You have a recording of a male and a female chatting with one another in an automobile. The recorder microphone is closest to the female and it is very hard to discern the male end of the discussion. The background noise is due to the high speed at which the automobile is traveling. However, there is one small sector of the recording where the car was stopped at a traffic light and both the male and female ends of the discussion are clear and discernable. To improve the clarity of the male voice during the times at which the car is traveling at high speeds, place the Spectral Filter in Spectral Copy mode. Highlight the male voice (only) at the point where the car was stopped at the traffic light and then click on the Sample Spectrum button in order to obtain a sound-print. Then, using this sample, Preview or Run the Spectral filter in the areas of the recording during which the male voice was not discernable.

Spectral Filter Application Example #3

You have a cockpit voice recording containing a lot of airplane noise. You desire to listen to hear if the flaps-down switch has been flipped at a certain point in time, but at the point in time in question, all you hear is random noise. You desire to enhance the playback of the recording to hear whether or not the sound of that switch being flipped by the pilot of the aircraft is present on the recording. You can go to a flight simulator of the aircraft in question and record the sound of that switch being flipped with a quiet ambient noise environment. Bring up that sound file and highlight the actual sound of the switch being flipped. Next place the Spectral Filter into Spectral Copy mode of operation and then click on the Sample Spectrum button. Bring up the cockpit voice recording wavefile and Preview it around the time period in question. This action should provide more gain of the sound-print spectrum occupied by the flipping of the switch potentially making it more audible. Obviously, you can't prove the negative by this technique, but if the switch is heard, you can demonstrate that something took place at the point in time in question. Of course, that positive result may represent the wrong switch being thrown, but that is beyond the scope of this application example.

Spectral Filter Application Example #4

You have recordings of an old television variety show from the 1950s (created before the advent of Video tape recording). Your job is to assemble a new digital edit of this television series. The dialog and music are both recorded on a cine optical track. However, a much higher quality copy of the musical portion of the show exists on magnetic tape (sans dialog). In the editing process, you need to use the optical track to assemble the edit, but you insert the magnetic track for the musical portion of the job because of its higher sonic quality. However, you observe that there is an unnatural transition between the dialog (optical) track and the musical (magnetic) track. You can create more natural transitions between the dialog and the musical interludes by employing the Spectral Difference mode of the Spectral Filter to the project. Use the Magnetic track as the “Source” Track and then apply that to the Target optical portions of the final edit using the Spectral Difference feature.

Spectral Filter Application Example #5

You have transferred an old acoustical recording of a famous opera singer to your hard drive. You have removed the noise, but the recording sounds honky, hollow, resonant and unnatural. But, you have a more modern recording of this singer created during the electrical recording period of time. You can use the Spectral Difference feature to create a more natural sounding result. Just use the more modern recording as the Source .wav file and then apply it to the acoustical recording as the Target.

Spectrograms

Both the DC8 and the DC Forensics Audio Laboratory versions include spectrogram displays (sometimes referred to as “spectrographs”). The DC8 version includes a Standard Definition system while the DC Forensics Audio Laboratory version includes a High Definition system in which the user can optimize and tradeoff between optimal frequency or time resolution displays. Since the High Definition Spectrogram is similar in overall operation to the High Definition system, it is recommended that the user should read the section pertaining to the Standard Definition system before proceeding to the High Definition section of this users guide.

View Spectrogram

A Spectrogram provides a method for displaying waveform data including Time, Frequency and Intensity all on the same graph. Time is represented on the horizontal (X) axis, frequency is represented on the vertical (Y) axis, while intensity (loudness) is represented by color or gray scale. The Spectrograph displays itself in the Destination window when this is checked. The Spectrogram is calculated for whatever waveform is displayed, highlighted and zoomed in on in the Source window and is time aligned with the same. What follows is a description of the Standard Definition Spectrogram found in the DC8 version of the product.

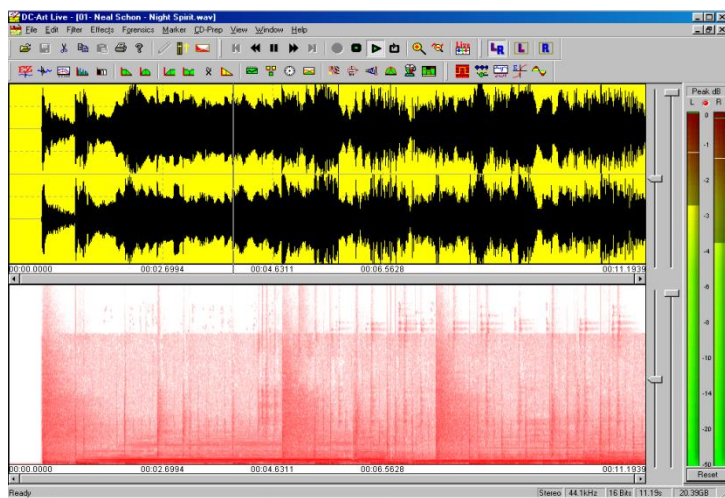


Figure 93 - The DC Forensics Spectrogram View

Display Controls:

- **Right Vertical Slider:** This slider sets the Maximum Amplitude Level (limit) used for the modulation of the spectrograms Z Axis. Its range of adjustment is from 0 dB down to as low as – 90 dB. Signals above its setting are suppressed. Note that this control operates differently compared to the Standard spectrogram's Right Vertical Slider.

- **Left Vertical Slider:** This slider sets the Minimum Amplitude Level (limit) used for the modulation of the spectrograms Z Axis. Its range of adjustment is from - 25 dB down to as low as - 125 dB. Signals below its setting are suppressed. Note that this control operates differently compared to the Standard spectrogram's Left Vertical Slider.
- **Zoom In*:** Allows you to zoom in on a portion of the .wav file as displayed in the Source window. After zooming is completed, the system will re-calculate the spectrogram for the zoomed-in segment displayed in the destination window.
- **Zoom Out*:** Allows you to zoom out on a portion of the .wav file as displayed in the Source Window. Similarly, the system will re-calculate the spectrogram for the zoomed-out displayed data in the destination window.

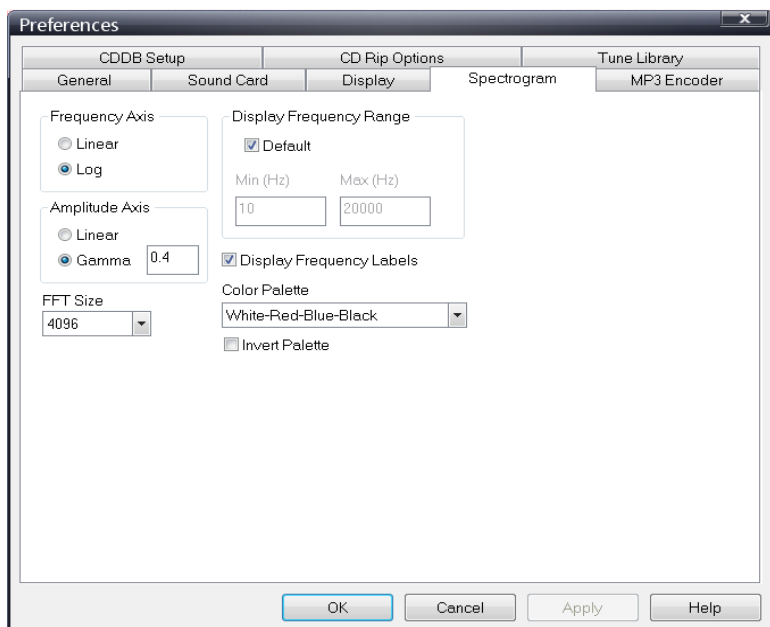


Figure 94 - The Spectrogram Display Preferences Menu

Display Preferences:

You can choose between a number of preferences associated with the spectrogram under the Preferences menu found under “edit,” or by left mouse - double clicking on the spectrograph display area. The following preferences are available to you:

- **Frequency Axis Selector:**
 - A. Linear
 - B. Log
- **Amplitude Axis (Z Axis or Chroma/Intensity Modulation)**
 - A. Linear
 - B. Gamma Scaling (Co-efficient of Non-linearity ranging from 0.1 to 10 with 1.0 being linear)
- **FFT Size:**

Choose between 32, 64, 128, 512, 1024, 2048, and 4096. Small values provide fast FFT update time, while large FFT sizes provide improved frequency resolution. The frequency resolution is the FFT size/2.
- **Color Palette:**

You have the choice of the following color gradients:

 1. Grayscale
 2. White to Blue
 3. White to Red
 4. White to Green
 5. White to Red to Blue
 6. White to Green to Red
 7. White to Red to Blue to Black
 8. White to Yellow to Red to Black
 9. White to Yellow to Green to Aqua to Blue
 10. Black to Blue to White
 11. Black to Green to White
- **Inverse Palette:**

This inverts the polarity of the video signal providing a different visual perspective of the spectrogram which sometimes is more revealing than the normal

polarity. For example, on the grayscale, black become white and white becomes black when the Inverse Palette checkbox is checked.

- **Display Frequency Range**

1. Enter Value for Minimum Frequency in Hz
2. Enter Value for Maximum Frequency in Hz which is limited to the file Sample Rate / 2.

- **Display Frequency Labels**

This feature turns the Frequency Labels along the Vertical axis On or Off

***Note:** For more information on methods for Zooming-In and Zooming-Out, please refer to that section of the User's Guide.

Important Note: The Sync files feature found under the View menu must be enabled so that the Spectrogram operates properly.

Spectrographs are useful for applications like spectrographic voice recognition or comparison (sometimes referred to as “voice-printing.”) Physiologically, speech is produced by the interaction of two mechanisms consisting of resonance and articulation. Resonance is produced by the nasal, pharyngeal and oral passages while articulators are produced by the jaw muscles, lips, teeth, tongue, and the soft palate. The human voice is acoustically modeled as a 4th order cascaded resonant system with an excitation signal called F0 (produced by the vocal cords). These acoustical signatures are referred to as formants. There are generally 5 formants that are identifiable starting with the fundamental which is usually designated as F0. Resonances produce formants designated as F1 through F4 are generally higher in frequency than the fundamental (F0). All of these formant frequencies lie somewhere below around 3000 Hz. F0 generally falls between around 70 Hz through around 270 Hz. Typically, audio samples that are around 2.5 seconds in length with the frequency display range showing information from 100 Hz to somewhere between 3 to 6 kHz are used for vocal comparisons. The so-called English “cue words,” often used for comparison are as follows:

{ The, To, And, Me, On, Is, You, I, It, A }

Here is a sentence that you can experiment with that incorporates all of the English cue words:

“It is important that I go to the bank on Friday to get a check for you and me”.

There are three sets of demo files in the Forensics 8 demo wavefiles directory which express the above English sentence. Three pairs of files were made for user testing. They include a male voice, a female voice and also that of a child’s voice. Each pair of files were recorded simultaneously; in each case one file was recorded through a very poor signal path while the other was recorded via a very high quality signal path. You can use these files in conjunction with the Spectrogram (and the Voice ID System) to study the differences between male and female voices expressing the same exact sentence. You can also study and compare those same voices as recorded by high and low quality recording systems. The files are as follows:

Female Voice ID Test Sentence - High Quality.wav

Male Voice ID Test Sentence - High Quality.wav

Female Child (12) Voice ID - High Quality.wav

Female Voice ID Test Sentence - Low Quality.wav

Male Voice ID Test Sentence - Low Quality.wav

Female Child (12) - Low Quality.wav

To perform a voice-print comparison, it is necessary to observe the same words contained on the two specimens (the display tile feature is helpful for this purpose). Your Diamond Cut Voice ID system is a useful tool for this purpose. You can use it to compare the vowel (generally F1 and F2) and consonant (generally F2, F3 and F4) formants of the human voice.

It is beneficial if the same recording equipment was used to record both samples of data to be compared, however impractical in most situations. It is also beneficial if the same ambient sound conditions are presented on the two samples. If there is a lot of background noise, consider applying one of the Speech filters before measuring the spectrogram. Speech filters can be found in the Band-pass filter preset menu and the Forensics Brick Wall filter preset menu. See those specific sections of the user’s guide for details. Lastly, the emotional state of the person(s) making the expressions should be similar. If one

specimen, for example, has the person screaming, and the other has the person sobbing or whispering, it will be difficult to draw any conclusions with a reasonable degree of certainty. Please note that spectrographic “voice-printing” is not admissible evidence in all court systems in the United States. Contact the court system or a legal expert in your area for details.

Note 1: Hot key access to the Spectrogram is available via **ALT+"S"**. It toggles between the Spectrogram Display and the software's Normal Display mode. Alternatively, you can use the “**Esc**” key to simply turn off the spectrogram.

Note 2: To print a spectrogram, use the Print commands found under the File Menu.

High Definition Spectrogram

The DC Forensics Audio Laboratory version includes a very High Definition Spectrogram which allows for FFT Sizes up to 131,072 (65,536 Frequency Bands). It also includes Decimation techniques to further optimize for higher Frequency Resolution as well as Zero-Pad techniques in order to optimize for better Time Interval Resolution. Both the frequency and time resolution parameters have been integrated into one simple and easy to use “Increase Resolution” slider control. Additionally, you can choose between several window techniques depending upon your specific needs. The overall combination of features makes the High Definition Spectrogram well suited to identify tape dubs based on multiple line frequency pickup signals. It is also useful for identifying edit points in forensics audio files. And, it provides an exceptional voice print display for detailed comparison between known and unknown voice sources. The High Resolution Spectrogram dialog box is pictured below:

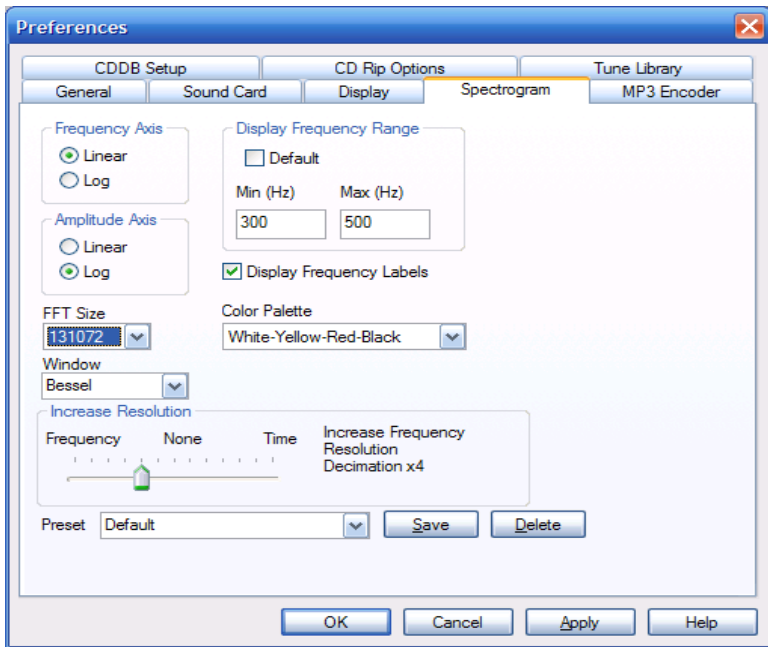


Figure 95 – The High Definition Spectrogram Dialog Box

Most of the functionality of the High Definition Spectrogram is the same as that of the Standard Definition version with the following exceptions:

- **FFT Size:** User Adjustable from 32 to 131,072
- **Window Choices:** Select between Bessel, Blackman, Hamming, Hanning, Kaiser 10, Kaiser 15, Kaiser 20, Rectangular, Triangular and Welch
- **Increase Resolution Control:** Moving the slider to the left increases the spectrograms Frequency resolution with the reduction of time resolution (by Decimation) while moving the slider to the right increases the spectrograms Time Interval resolution (by Zero Padding) with a loss of frequency resolution. You make the tradeoff that you require using the Increase Resolution Control. The Decimation and Zero Padding values are annotated just to the right hand side of the

slider control. In the middle position, neither Decimation or Zero Padding are applied to the system.

- **Point and Click Measurement:** You can point your mouse to any point on the High Definition Spectrogram and single-left click it and it will display the Frequency in Hz and the Amplitude in dB at that location. The exact measurement location is indicated by a small crosshatch (+) sign on the graphical display adjacent to the numeric display.
- **Presets:** Several factory presets are provided with the software. You can also add your own favorite presets by using the Save command button found in the Spectrogram dialog box. The delete button is used to eliminate unwanted presets from the preset listing.

Note: The Spectrogram Dialog Box can be accessed by either double-left mouse clicking on the spectrogram display itself or via the Preferences Menu found under “Edit”.

Voice ID

Your Diamond Cut Forensics Audio Laboratory software contains a special feature called “Voice ID” to help you identify and rank the speech formants* of a highlighted cue word or verbal expression. It also includes the ability to display the frequency response, power cepstrum,* and complex cepstrum* graphs of the highlighted signal simultaneously.

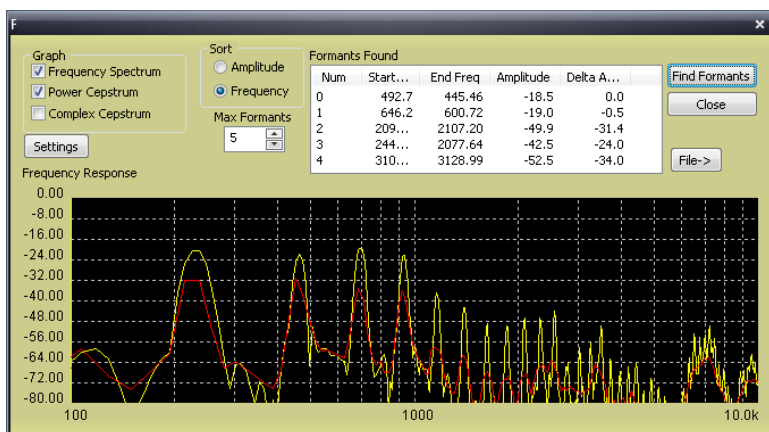


Figure 96 – The Voice ID Dialog Box

This feature works in conjunction with your high resolution spectrograph or your Direct Spectral Editor (DSE) found under the Edit menu. The Voice ID feature allows you to highlight a cue word or phrase and then (after activating the Spectrogram or the DSE) it finds and automatically ranks the various formants contained in that portion of the waveform. It also provides you with a frequency response and cepstrum graph within the Voice ID dialog box. The system can measure the formants and cepstrum for time intervals up to 5 seconds based on the highlighted sector of the file. Most useful voice vowel formant identifications are performed on a much smaller range of time, usually in the 40-100 mSec range.

You can choose the number of formants that you want to be identified and ranked. Formants are always sorted based on the highest average amplitude of the formant. You can choose between having these results displayed in an Amplitude or Frequency priority order. You can also choose which formant number you want the ranking to commence with by choosing between “Start at F1” or “Start at F0”. Maximum and Minimum formant frequencies can also be defined by these features found under the “Settings” dialog box.

The system will generally show the first formant as F0 on the spectrograph (and Num 0 in the table of values). This F0 represents the fundamental excitation frequency. The component signals (formants)

that comprise a cue word or phrase are displayed in terms of start and stop frequency, amplitude and start time by Fn.

Sometimes, you will encounter noisy files which can confuse the Voice ID System. In these cases, it may be useful to limit the range of frequencies used by the system or to raise the “Spectral Smoothing” value. By default the Voice ID frequency range is set to a lower limit of 100 Hz and an upper limit of 5,500 Hz. You can access and change the various internal parameters of the Voice ID system by using the Settings button on the Voice ID dialog.

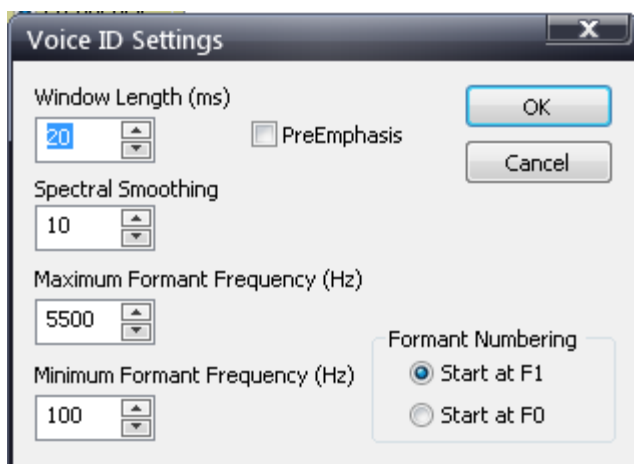


Figure 97 – The Voice ID Settings Dialog Box

Note that that the Voice ID is not affected by the DSE (Direct Spectral Editor) frequency selections or the range of displayed frequencies in the spectrograph.

It is recommended that files should have a sample rate of at least 22 kHz, with 44.1kHz being a better choice. If necessary, you can convert the file using the file “Change Sample Rate/Resolution” feature found under the Edit Menu.

You can adjust some of the internal parameters associated with the Voice ID function by clicking on the “settings” button. The Voice ID Settings Dialog Box will appear giving you control over the time

window aperture, the spectral smoothing degree, the maximum formant frequency sought after and the application of pre-emphasis. The Window length is basically the signal frequency / time parameter of the Voice ID system. The most common value used is 20 mSec; sometimes it may be useful to try other values depending on the length of the phoneme on interest.

The Pre-Emphasis option is available via the settings dialog. This applies a +1 slope (+6 dB/Octave) from 75 Hz to 5,000 Hz to the signal being analyzed. This is used to compensate the natural rolloff of the vocal tract and flatten the vocal formant spectrum.

After the “Find Formants” has been “clicked” and the calculations have been completed, each formant and its trajectory is displayed on the spectrograph as a number displayed in a rectangular box. They are annotated with both time and frequency coordinates that correspond to the values displayed in the Voice ID dialog box (“Num” column in the table of values). “Start” and “End” frequency values for each formant are annotated in the table of values. Average amplitude values for each formant are given in dB relative to 0 dB which is the maximum value that can be displayed. Also, a column called “Delta Amp” (Delta Amplitude) displays the various formant amplitudes normalized to 0 dB for easier comparison to a reference cue word or phrase.

All of these data can be exported to a text file so that further analysis can be performed with such programs as an Excel or other equivalent data analysis systems. The data that is exported is the frequency and time values for each of the formant tracks.

The software supports two extensions, .txt and .csv, and they both provide you with the same exportable data. Csv (comma-separated values) files are comma delimited while .txt files are tab delimited. To export the data, just click on the “File” button, then select “Export to a file”. You can set the file path and extension to go to the directory of your choice. You can also choose “copy to clipboard” if you want to bypass writing the information to a file and copy it directly to another program.

Two cepstrum graphs and a frequency response plot are also provided with the Voice ID feature. The highlighted portion of the file’s

Frequency response shows the relative amplitude in dB plotted vs. frequency while the power and/or complex cepstrum can be simultaneously displayed in terms of amplitude vs. quefrency*. Use the “Graph” checkboxes to select the desired graphing mode. The Frequency Spectrum will be drawn in Yellow, the Power Cepstrum in Red and the Complex Spectrum will be drawn in White. You can select any and/or all of these graphical modes depending on your needs. The smoothing control applies to these graphs with higher values producing higher levels of graphic smoothing. The smoothing scaling factor runs from 0 (which produces no smoothing) to 20 (which results in the maximum degree of smoothing). This smoothing control also affects the formants detection functionality.

The Voice ID System Operating Procedure(s)

1. Bring up the file of interest in the time domain display.
2. If it is not 44.1 kHz, use the change sample rate feature to change it to 44.1 kHz. This feature is found under the Edit menu.
3. Optionally, go to the “View” menu and bring up the “Time Display” which makes it easier to see your highlighted “Span” time.
4. Go to the “Forensics” Menu and click on the “View Spectrogram” item.
5. Optionally modify the spectrograph properties by using the right mouse button menu to Edit the spectrograph properties.
6. Listen to the file and then highlight the cue word or phrase of interest on the spectrogram.
7. Click on the “Voice ID” function found under the Forensics Menu.
8. Set up the various “Voice ID” parameters to your preference. Generally, the Sort should be set for “Frequency” and the Max Formants would generally be set for 5 (which will allow the system to identify F0 through F4).
9. If you are interested in Frequency Spectrum or Cepstrum graphs, check the appropriate checkboxes in the “Voice ID” dialog box. If not, leave all checkboxes unchecked.
10. Last, click on the “Find Formants” button in the “Voice ID” dialog box.
11. The system will then calculate the various Formants and display their various numerical attributes in the table of values within the “Voice ID” dialog box.

12. Please be aware that the voice formant frequencies are very sensitive to the exact location of the selected speech. Small variations in time can cause different formants to be found. Likewise the window length time (under the settings dialog) will affect the type of formants found. Typical analysis is done with a 20 ms window (aperture).

12. The trajectories of the various Formants will appear on the Spectrogram along with their numerical labels.

13. If you want to analyze this data statistically or in any other way, you can click on the File->Export button and a file will be created which can be used external to your Diamond Cut software.

*Note 1: Definitions of Power Cepstrum, Complex Cepstrum, Formants and Quefrency can be found in the glossary section of this documentation.

Note 2: The Voice ID dialog box is user sizable; just use your mouse to drag the box margins to create the size that you desire.

Note 3: To print the Frequency Response and/or Cepstrum graphs, use the Alt Print Screen command and it will be recorded on your system clipboard to be used as needed.

DeClipper



The De-Clipper is provided to reduce the distortion which results from an over-driven or clipped signal. When viewed in the time domain via the Source display window, this problem often looks like the signal has a “crew cut” or is “maxed-out.” It is equipped with two modes of interpolation, one having adjustable strength while the other uses a complex frequency domain method of interpolation. The “Adjustable Strength” mode is most useful on signals having a fairly high level of coherence while the “non Adjustable Strength” mode is best suited to signals having a higher level of randomness in their makeup. The “non Adjustable Strength” routine requires a high level of CPU and thus is quite slow, whereas the “Adjustable Strength” mode is much faster. Experimentation is the best way to find the optimal mode to apply to your particular clipped file. We recommend starting with the Adjustable Strength mode first, since it contains the fastest running replacement algorithm.

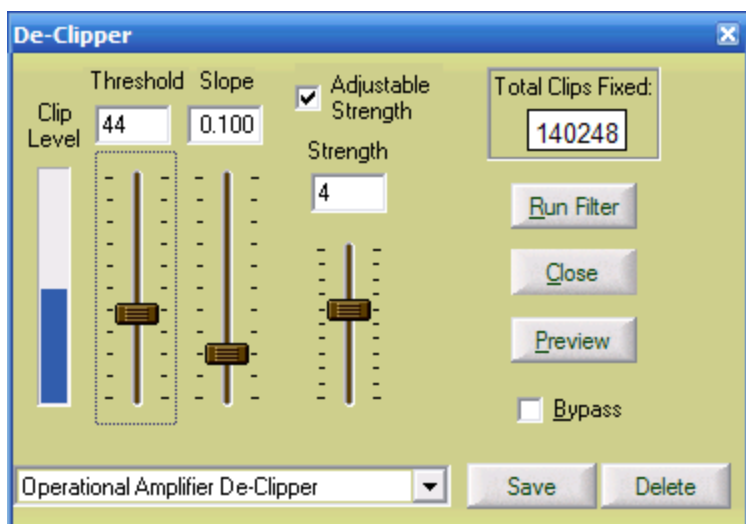


Figure 98 - The DeClipper

The De-Clipper can be used to repair signals, which were either clipped by digital or analog mechanisms. It performs its magic by detecting signals with very low or zero values of slope (user adjustable from 0 - 0.5) above a settable threshold amplitude value. When this condition is detected, the routine mathematically interpolates a new signal and replaces the zero slope portion of the bad waveform with one containing curvature. This results in decreased distortion. If the material being de-clipped has been directly clipped by the digital recording process (in other words, the signal is clipped at full scale output as indicated on the destination window), then you must first decrease the overall gain of the .wav file by 6 dB before applying the De-Clipper. This sort of de-clipping can be accomplished with very low values of slope. If the signal was clipped previous to the transfer to the digital domain by an overloaded analog amplifier, the signal can be de-clipped by raising the slope control until the “total clips fixed” display starts incrementing. The following is a listing of controls available on the de-clipper:

- **Adjustable Strength Checkbox**

This checkbox determines which of two different replacement algorithms are used by your De-Clipper. Check this box when

dealing with audio signals which are fairly coherent in nature and un-check this box for signals which are more random (or stochastic) in general nature.

- **Threshold**

The threshold determines the amplitude above which signals will be applied to the de-clipper detector. Lower values imply a more aggressive response. The range for this control is from 10 to 100 on a relative and normalized percentage scale.

- **Clip Level**

The clipped level of your signal is measured and displayed on this bar graph and is useful for setting the “Threshold” control. Close alignment of the “Threshold” control with the “Clip Level” while watching the “Total Clips Fixed” display will allow you to find the best “Threshold” setting. The “Total Clips Fixed” indicator will begin to increment when the proper threshold is found. Do not overdrive the system by setting the “Threshold” control to an excessively low setting as this may introduce distortion into the process.

- **Slope**

The Slope control determines the “flatness” of the clipped waveform that will be interpreted as a “clipped” event. Low numbers like zero, imply a perfectly flat line as will be found in digital clipping. Higher numbers represent slight slopes associated with analog clipping. The range for this control is from 0.000 to 0.500 with 0 representing zero slope and 0.500 representing a 45 degree slope.

- **Strength**

The Strength control affects the curvature of the applied interpolated waveform. Adjust this for the best sounding (or looking) replacement waveform. The range for this is from 1 to 5, with 5 being the most aggressive.

- **Total Clips Fixed:**

This is a numerical display indicating how many “clips” were detected and fixed by the de-clipper. It is useful when using

“preview” mode to assure that the routine is detecting the clipping while adjusting the various controls.

- **Bypass**

This bypasses the de-clipper so that you can preview and hear the before and after results of the de-clipping process in an instant.

De-Clipper Operating Procedure (Tutorial)

File Preparation: Before de-clipping a .wav file, it is necessary to reduce its amplitude by 6 dB before applying the following procedure. To reduce the gain of the .wav file, use the Gain Change feature found under the Edit menu. The reason for this step is to provide the de-clipper algorithm enough headroom to interpolate the clipped peaks of the .wav file.

1. Set the Threshold control to its maximum value (all the way up).
2. Set the Slope to a value close to zero for digital clipping, but higher for analog clipping.
3. Preview the filter and watch the Total Clips Fixed display.
4. Reduce the level of the Threshold until it begins to show increments on the Total Clips Fixed display.
5. Adjust the Strength for a minimization of distortion as heard in preview mode.
6. Run the Filter.
7. Done

Note: You can look at the results produced by the De-clipper after you have run the Filter. You will notice that the flat-topping has been replaced with smooth rounded waveforms if the controls have been set properly.

DSS Dynamic Spectral Subtraction™ **(Forensics Version Only)**



The DSS feature in the DC Forensics software is a unique and powerful tool capable of recovering speech from recordings containing loud music or other coherent noise. Until now, a recording that was covered

or masked by loud music was basically a lost cause. DSS decoding is designed to make it possible to attenuate this ambient music and uncover the speech.

Basic Approach of the DSS Filter

DSS works by performing a continuous and intelligent subtraction of one audio signal from another. Normally, with a Forensic recording containing speech masked by loud audio, you will require a reference recording containing just the audio that needs to be removed. The audio track containing the music or other audio to be removed is called the “Reference Track.”

The DSS Controls

The following four controls are active when operating in DSS mode.

1. Selection Box:
 - A. Mono DSS – Delay Reference
 - B. Binaural DSS – Left Reference
 - C. Binaural DSS – Right Reference
2. Attenuation: Range = 0 to 100 (Tune for a Null in the Noise) Null usually occurs around 50.
3. Channel Time Offset: (Adjust for the best nullification of the unwanted signal).
 - A. Horizontal Slider Control: Course Time Offset Adjustment
 - B. Rotary Dial Control: Fine Time Offset Adjustment
 - C. Offset Samples Display Window (given in # of samples)
 - D. Offset Time Display Window (given in mSec)
4. Advance %: Range = 10% to 50% (Tune for minimum digital artifact production). In most instances a setting of around 40% is effective.
5. FFT Size: 256 to 8192 FFTs (adjust for best unwanted signal rejection)
6. Output: 0 to 20 dB – Used to compensate for gain loss when the “sweet – spot” is found when using the DSS.
7. Red Clip LED: Adjust the Output Control downwards when this indicator lights so that clipping distortion does not occur.

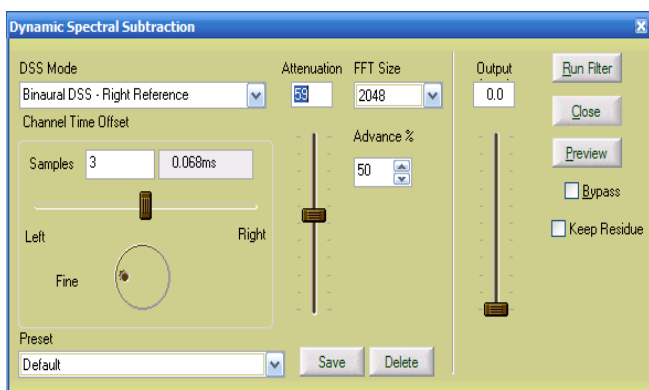


Figure 99 - DSS User Interface

Recording a Reference Track

There are many ways that you can obtain a reference track. These examples should make this clear:

Real Time Methods*

1. Place two microphones in the venue. Place one near the target conversation and place the other near the source of the background audio source such as a TV, stereo system, jukebox, or a live band. Record these two signals with a stereo tape recorder or computer.
2. Wire the investigator with two microphones. Place one near the investigator's chest and place the other much lower on the investigators body, like down in his or her sock or shoe. Record both signals with a miniature stereo tape recorder.
3. Wire the room with a wireless microphone located near the sound source like the TV, Stereo, jukebox or live band. Wire the investigator with a wireless microphone located near his or her chest. Record both signals with a remote stereo recorder or computer.

Non Real Time Methods* ‡

1. Assume that you have a recording made in a bar or similar venue that was recorded with a monophonic pocket tape recorder. The jukebox or other interfering music source is

covering over the targeted speech. You can go back later with the same recorder and record the same exact song that was being played. This will become your reference track for DSS decoding.

2. You have the same situation as stated above, but you have a second tape recorder on site that is recording only the noisy background environment.
3. You have the same situation as stated above, but you record the same music that had been playing at the venue from a commercial audio CD. This process can be performed in non real time back in your audio lab.

*** Note 1:** Digital Recorders produce better results than Analog recorders in DSS decoding applications due to their crystal controlled speed regulation. This may not be the case, however if the digital recorder uses “lossy” compression.

‡ Note 2: If the interfering source of audio was a radio or television, many broadcast stations maintain an archive of “air-checks.” You may be able to access the required broadcast “air-check” recording through either the use of diplomacy or a court order.

Obtaining a reference recording is an important step in removing loud coherent noise sources such as music. Using the Real Time Methods described above, technique number 1 (there under) will produce the best results. In the Non Real Time methods described above, number 2 (there under) will produce the best results since it will rely on a reference recording that closely resembles the noises that you will be attempting to remove from the target signal.

Modes of DSS Operation

There are 3 DSS modes of operation available in the product. To select one, drop the DSS Mode Selector box. As you see, you can select either the right or the left channel as the reference track.

If you have no reference track recording and cannot re-create one, you can try to use the setting called Mono DSS Delay Reference. This will attempt to attenuate the noise by comparing the audio at an instantaneous point in time and comparing it with a point at some other

time before or after the comparison point. This has the effect of allowing the program to create its own reference signal.

Note: This method is inferior in comparison to any method utilizing a true reference track.

Creating a Stereo track from two discrete tracks in non real time situations: *

The audio file that you will actually clean up using DSS decoding is ideally going to be a stereo file that you recorded in real time. However, often that is inconvenient and non real time methods must be used. In these cases, one channel of the file will be the recording with the speech you want to recover (the Forensic recording) and the other channel will contain just the music or other non-random audio. These two recordings will have to be combined into a single stereo (binaural) recording. The easiest way to accomplish this is to use the File Split and Re-Combine function under the Edit menu. Here's the procedure:

1. Take the two recordings (the Forensic recording and the reference recording) and convert both of them into monophonic files if necessary by using the File Converter Filter.
2. Use the File Split and Re-combine feature to merge these two mono files into a stereo (binaural) file
3. Time align these two files by either cutting a piece from the beginning of one of them or insert a piece of silence of appropriate length in front of one of them. Note: Using the Markers and the Time Display feature is quite helpful to precisely measure the time displacement between tracks to calculate how much audio must either be cut or inserted to result in the proper time alignment. The two tracks should be time aligned to within +/- 25 milliseconds of each other for optimum results.

*** Note:** If the interfering audio came from a live performance, having the live performance re-created by the talent after the fact will not produce a useable reference track for DSS decoding.

DSS Adjustments:

The controls that are active in the DSS filter are Attenuation, FFT Size and Delay. The Attenuation setting will control the amount of noise reduction that is performed by the DSS filter. You can think of this control as being analogous to balancing the weight(s) on a balance scale. Moving it up will reduce the noise more until you pass through a “null” point in the background music. You need to tune the attenuation control for the most music reduction, which generally will occur around an Attenuator setting of 50, as long as both discrete channels are relatively balanced in amplitude with respect to one another.

The FFT size controls the size of the frequency “buckets” that are being used internally by the filter. Smaller numbers allows for more “self adjustment” of the filter to the mismatched forensic and reference recordings. Larger values produce better frequency discrimination and overall attenuation. We find that settings of 1024 or 2048 generally produce good overall results, but smaller or larger settings should be tried as well.

The Advance % control is generally set to 50%. However, it is worthwhile experimenting with other values of Overlap in order to minimize the introduction of digital artifacts into the final resultant signal.

The Time Offset control can also help with time alignment mismatched audio channels. You can calculate the actual delay time between the reference microphone and the target microphone in milliseconds by applying the following formulae:

$$TD = (\text{Delay Setting} - 1)(\text{FFT Size}) (\text{Advance \%} \times 0.01) / \text{Sample Rate}$$

wherein:

Delay Setting is an Integer value from 1 to 10

&

Overlap is a value from 10% to 50%

&

Sampling Rate is a value given in Hertz

&

TD is the resultant delay time given in milliseconds

After adjusting the Channel Time Offset for a minimization of background music or noise, you can ascertain the distance between the reference microphone and the target microphone by looking at the Offset Time Display Window. Each mSec represents about 1.1 feet of distance between the two. As always, simply preview the audio and make your adjustments in the DSS filter window and also with the Time Offset slider in the File Conversion filter.

Primary Compensation Issue with the DSS Filter in Real Time Situations:

The primary problem encountered when using the DSS filter in real time applications arises from the distance between the two microphones used to make the binaural recording. Because a physical distance in the venue separated the two microphones, the propagation delay (sometimes referred to as group delay) of the signal between the two microphones in the room may need to be compensated for. Since sound travels at 1131 feet / second at 70 degrees F (or 1.131 feet / millisecond), large distances between microphones can cause misalignment between the two tracks of the binaural file. In real time situations, the noise signal on the Target Track will lag the Reference Track at the rate of 0.884 milliseconds per foot. The File Conversion Filter and its Time Offset control can be used to compensate for up to 20 milliseconds of propagation delay representing a distance between the microphones of up to about 23 feet. If more distance had existed between the two microphones, multiple File Conversion Filters can be cascaded in the Multi-Filter to increase the total compensation time.

Primary Compensation Issue with the DSS Filter in Non Real Time Situations:

The primary difficulty encountered when using the DSS filter in non-real time applications arises from the lack of acoustical matching between the Target Forensics recording and the re-created Reference

Track. In other words, room resonance, frequency response or natural room reverb may not exactly match your re-created Reference Track. Performing some pre-processing on your Reference Track can compensate for these acoustical mismatches. You can use the 20-band equalizer to match the resonance and frequency response of the room. Also, you can use the Reverb to simulate the acoustical reflection characteristics of the venue. These steps will rely on your own sense of hearing to create the match. When you listen to the music on the Target Forensics recording, focus your listening on its musical content. Then try to create that same sound on your re-created reference track using the above-mentioned tools. Then use this pre-processed track as your final Reference Track to be applied to the DSS filter.

Dynamic Spectral Subtraction™ and DSS™ are Trademarks of Diamond Cut Productions, Inc. 2003

Cell Phone Noise Filter (Forensic Version Only)

Some cell phone systems can interfere with audio equipment despite their use of signals well outside of the audio spectrum. Their carriers operate in the Giga Hertz range (0.850 – 1.9 GHz). GSM (Global System for Mobile Communications) based cell phones send out their packets of data in short bursts. These RF (Radio Frequency) bursts (pulses) can be radiated into audio system front end amplification devices like bipolar junction transistors or field effect transistors found discretely or inside operational amplifiers. The non-linear nature of these devices coupled with the integrating effect of collector to base (in BJT based circuits) or drain to gate (in FET based circuits) Miller capacitance creates a parasitic AM (Amplitude Modulation) demodulator circuit. Thus, the envelopes of these RF bursts of energy are de-modulated into the audio range of frequencies. The repetition rate of these RF bursts lie within the audio spectrum. Thus, they can be detected and heard in an audio signal chain that is not sufficiently shielded. Because these noise bursts become audible they can render an audio signal pathway extremely noisy, contaminated with a “buzz” like sound and thus, often unintelligible. The Diamond Cut Cell Phone Noise Filter is designed to attenuate the noise found in this situation.

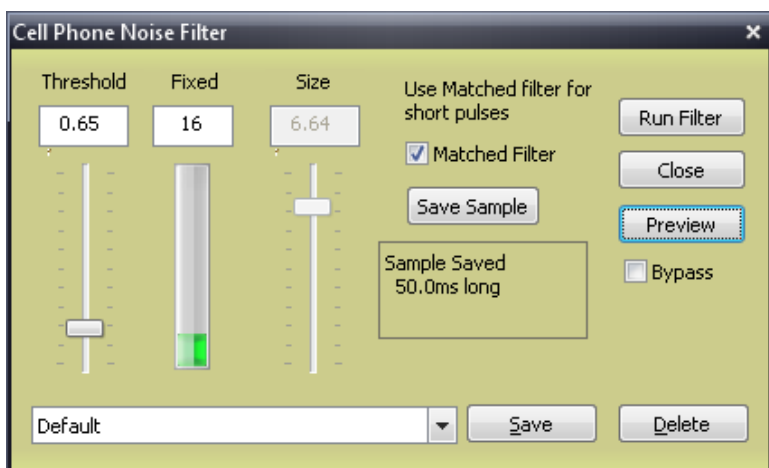


Figure 100 – Cell Phone Noise Filter

The filter has two modes of operation. The first and more basic mode is more automatic, but often less effective while the second mode (Matched Filter) is more discriminating. When the Matched filter is not checked, the system automatically attempts to find the cell phone noise pulses and interpolate them out of the audio signal. When the “Matched Filter” box is checked, you will need to highlight a sample of one of the noise pulses in the time domain display of the software and then click on “Save Sample”. The system will then use a time domain technique of pattern matching to detect further pulses in your audio stream and then replace them with interpolated audio.

Automatic (Non-Matched Filter) Mode

Un-check the “Matched Filter” checkbox. Click on the Preview button and then adjust the ratio control until the pulses are detected and reduced in amplitude. Then, adjust the “Size” control for the best overall signal intelligibility. Size relates to the length of the noise burst and not its amplitude.

Matched Filter Mode

Check the checkbox labeled “Matched Filter” Zoom-in on a chain of cell phone pulses in the time domain display. Then, highlight a single pulse event making sure to capture the entire event. Do not highlight a train of pulses, just a single impulsive event. Be sure not to highlight any of the voice signal surrounding the noise pulse or burst. Next, click on the “Save Sample” button of the Cell Phone Noise Filter dialog box. Zoom back out on the time domain and then hit “Preview”. Adjust the Ratio for the best level of detection (noise reduction). Note that the “Avg Time” control is not active in Matched Filter Mode.

Note: This filter works best after your file has been up-sampled to 44.1 kHz, 16 bits.

Auto Voice Filter (Forensic Version Only)

The Auto Voice Filter is an adaptive system optimized for separating voice signals from random noise automatically. It uses two independent mathematical processes that work in harmony with one another in order to perform its job. One process identifies statistical noise while the other process identifies the human voice component of a signal. The Auto Voice Filter uses both of those pieces of information to separate the voice from the noise using sort of a “push-pull” methodology. The Auto Voice filter will continually adapt itself to varying signal to noise ratio conditions. To operate this filter, merely adjust the “Attenuation” slider control for the optimal signal quality while previewing. The optimum quality will be found as the best balance between noise reduction and speech intelligibility. Moving the attenuation slider upwards raises the aggressiveness of the filter and vice versa. Adjust the “Output Level” control such that the “Overload” LED does not light up on transient signals in order to minimize the creation of distortion. When you have discovered the best settings, highlight the entire sector of the file in need of filtering and then click “Run”. As with other Diamond Cut filters, you can save your favorite presets for future recall. This filter is limited to files using sample rates between 22.05 kHz to 48 kHz. If your file uses a different sampling rate, convert it to 44.1 kHz first before attempting to use the Auto Voice Filter. It works best with 16 bit resolution files. The Auto Voice Filter is monophonic only. Stereophonic files are

summed to monophonic first and then processed. The resultant processed output file is presented in a dual-channel monophonic format.

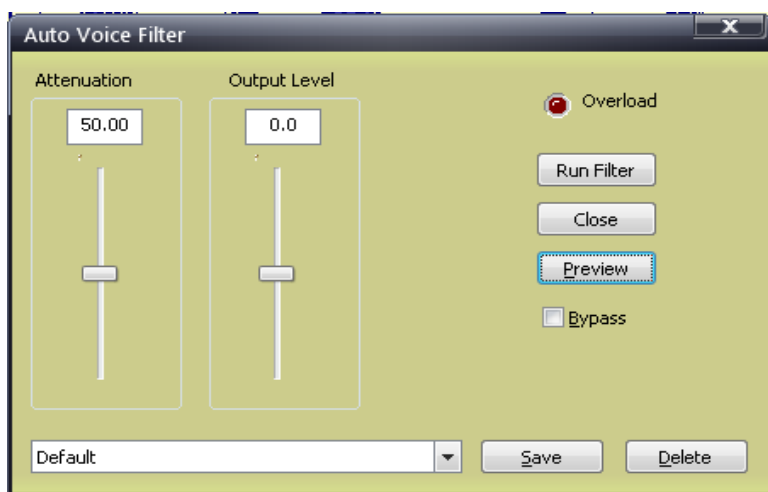


Figure 101 – The Auto Voice Filter

Note: This filter is optimized for Forensic Audio only; as such, it is not what one might call a “High Fidelity” system.

Voice Garbler (Forensic Version Only)

Sometimes it is necessary to disguise a person’s voice in order not to break cover or to protect a person’s anonymity that may be providing certain types of legal testimony. Your Diamond Cut Forensics Audio Laboratory software includes a Voice Garbler (voice disguiser) for that purpose.

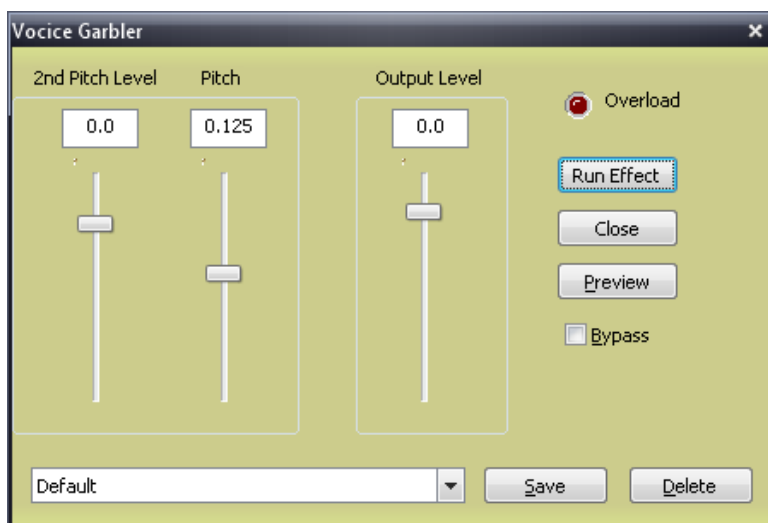


Figure 102 – Voice Garbler

The algorithm used renders the garbled voice almost impossible to reverse when the controls are set properly. This distinguishes this “garbler” from others that you may have worked with. To help maintain its security, we are not going to reveal anything about the method of operation of this algorithm. As a matter of fact, we are not going to use accurately descriptive naming conventions for the primary controls. They are simply “Pitch” and “2nd Pitch Level”. There are no units assigned to the numeric values associated with these controls either. To operate the Voice Garbler, simply adjust “Pitch” and “2nd Pitch” until you achieve the disguised voice effect that you desire. Switch the “Bypass” checkbox on and off as you vary the controls until the persons voice can no longer be uniquely distinguishable while maintaining the intelligibility of the content of the speech. The only accurately descriptive control is the “Output Level”. Adjust this so that the Clip LED does not illuminate if you are concerned about output signal clipping. As with other Diamond Cut filters and/or effects, you can save your settings by clicking on the Voice Garbler “Save” button.

Note: The Voice Garbler is not a high-fidelity system. Glitches (discontinuities in the signal) are a normal byproduct of its method of operation and are an element of its non-reversibility.

Additional Forensics Features

There are additional tools that will be of value for forensics work which are not found in the Forensics menu. Some of them are as follows:

1. Automatic Level Control (ALC or AGC) (Dynamics Processor):

Sometimes, these algorithms or systems are referred to as Automatic Gain Controls or AGC's. This feature provides upward expansion of signals below the threshold line and downward compression of signals above the same threshold. This feature is useful in forensics applications where there is a large variation in signal levels between several different parties that may be communicating with one another. It is also useful for the broadcast of live sporting events (if you have the DC FORENSICS version of the product) in which the crowd reaction is of interest when the announcer is not speaking. Simply clicking on the ALC box in the Dynamics Processor activates this feature. The threshold, attack, and release controls are still active when this function is invoked. Many Forensics specialists have also found the automatic level control feature (ALC) in the Dynamics Processor to be very useful in their work. This feature is particularly useful for correcting near-party / far-party loudness discrepancies of recorded phone conversations and other situations where signals vary greatly from time to time.

2. Punch and Crunch Effect operating in Narrow band mode:

This is useful for improving the intelligibility of signals having very small values of dynamic range in the speech portion of the spectrum. It is advised that a speech filter is run first before applying this filter for enhancement. For more information, please refer to the section on the Punch and Crunch effect.

3. Continuous Noise Filter:

This can be very useful for decreasing "in-band" noise found on Forensics recordings. A noise fingerprint will have to be taken from a section of the recording having only noise and no signal in order to use this filter. For more details, refer to the section on the Continuous Noise Filter.

4. Harmonic Reject Filter:

This is very useful for attenuating buzzes found on some recorded phone conversations. Several passes through this filter are often required for severe noise situations. Also, it is worth trying both odd and even harmonics of the fundamental buzz frequency. The fundamental buzz frequency can be found by using the Spectrum Analyzer located under the “View” menu.

5. Impulse Filter:

This is very useful for eliminating static from body mike recordings transmitted via an RF carrier.

6. Slot (which is a subset of the Notch) Filter:

This can be useful for isolating specific sounds buried in a recording.

7. Median Filter:

This is useful for bringing out the subtle articulations of speech in low-bandwidth or muffled voice recordings, especially when used in conjunction with its “Weighting” function.

8. Spectrum Analyzer in “High Resolution” mode:

This is useful for tape authenticity verification and other forensics analysis functions.

9. Source Time Display window in conjunction with the Timer found in the View Menu and the Markers found in the Marker Menu:

This is useful for precisely measuring the time between multiple events such as gunshots.

10. Live Mode is useful for surveillance situations in which signals must be cleaned up “real time.”

11. Live Log to Disk Mode: This is useful for recording conversations selectively while in Live, real time mode.

12. High Precision Spectrum Analyzer (Please refer to the section dedicated to this feature for details.)

- 13. 30 Band IIR Based Graphic EQ** (Please refer to the section dedicated to this feature for details.)
- 14. Forensics Adaptive Frequency Domain Filter (Forensics AFDF)** This is the twin sister of the “Adaptive Filter” found in the Forensics menu, but uses Frequency Domain rather than Time Domain techniques to achieve its end. For details, please refer to the section in the CNF description dedicated to this function.
- 15. Spectral Inverse Filter** – This is useful for improving the intelligibility of muted or muffled recordings and is part of the Spectral Filter.
- 16. Cell Phone Noise Interference Attenuator** – This is useful to reduce cell phone interference on electronic recording electronics and is found as a set of presets in the DC Forensics Multifilter.
- 17. Overtone Synthesizer** – The Overtone Synthesizer in the Forensics version of this software uses a different set of internal filters and ranges compared to the standard version making this effect more useful for enhancing the intelligibility of garbled speech. Its range of frequency settings overlap that of the standard version (DC8) and so it will do everything that the standard version does and more. The lower limit of the frequency setting is lowered from 3,500 Hz down to 2,000 Hz.

The Marker Menu

DC8/DC FORENSICS provides you with movable red timing markers that can be placed within either the Source or Destination workspace. They are the .wav file equivalents to bookmarks. You will find them to be useful tools when utilizing such editing features as the Copy, Cut, Mute, etc. The Marker menu is also complimentary to the CD Prep menu. It is very useful when you need to take a very long .wav file and Chop it up into smaller files manually for indexing onto a CD-R. The following commands are provided under the Marker Menu:

Add Markers:

This feature adds up to 100 markers to the desired Workspace. Clicking here will drop a marker at the beginning of a selected area. If no area is selected, by default, the marker is dropped at the far left of the .wav file. This command can also be accomplished by use of the right click menu.

Clear All Markers:

This feature erases all existing markers.

Highlight Marked Area:

This feature highlights in Yellow the area of a .wav file located between two previously defined red markers.

Drop A Marker:

This command allows you to drop a marker during playback so you can audition your audio while “marking it up” for in-depth study later. Simply hitting the “M” key on your keyboard during playback can also drop markers.

Go To Next Marker:

Exactly as the name implies...allows you to quickly navigate to the next marker on the right. This can also be accomplished by simply pressing the “N” key on your keyboard.

Go To Previous Marker:

Same as above, only this command takes you to the previous marker on the left. This can also be accomplished by simply pressing the Shift + “N” keys on your keyboard.

Re-Number Markers:

If your markers have become out of order, you can have the system automatically re-order them numerically by clicking on this function. This is useful if you are going to use the marker numbers as individual track number markers.

Label Marker:

By default, DC8/DC FORENSICS just places a number on each marker. If you use several markers, it may be necessary to label them more descriptively. Select the marker you want to label by left clicking on it (it will change from red to black). The Label Marker command allows you to be more precise with your markers; right clicking with your mouse and activating the Label Marker command can also accomplish this same task.

Lock Markers:

To lock the position of your markers, simply click on this menu item and your markers cannot be accidentally moved while you're working on your file. In order to move them, you'll need to press the Ctrl button plus drag the mouse. You can still remove markers if desired; this feature is a positional lock, not a marker erasure protection feature.

Merge Source Markers into Destination:

There are situations where it may be desirable to transmit the marker locations from a Source file into a Destination file when working in Classic Edit Mode. Clicking on this feature will facilitate this operation when markers are present in a Source File and a file of equal length is present in the Destination time display window.

The CD Prep Menu

This menu contains the tools that help you prepare for the final stages of your restoration project. The CD Prep Menu does everything but copy your files to CD.

Quantize for CD Audio

This feature moves a marker to a multiple of 2,352 Bytes to provide computability with CD data grouping so that glitch-less indexing can occur. This feature is particularly useful when chopping a large (continuous concert type) .wav file into pieces for transfer to CD-R. If you have a .wav file open, merely click on this menu item and the file will be properly quantized.

Using Quantize on a Single File (Tutorial)

1. Bring up desired .wav file in the source workspace
2. Click on the CD prep menu item.
3. Click on Quantize for CD Audio

Using Quantize and Chop File on Large File (Tutorial)

1. Bring up desired .wav file in the source workspace
2. Place markers at all of the locations at which you desire to break the file into individual .wav files. These are usually placed at the silent portions of a recording, between cuts.
3. Click on the CD prep menu item.
4. Click on Quantize for CD Audio. All of the markers shall be moved into the proper positions for CD quantization.
5. Chop file into pieces using the command by the same name.

Chop File into Pieces

This command breaks long .wav files into smaller .wav files as defined by the locations of your various markers. This command will remain grayed out until your file has markers laid.

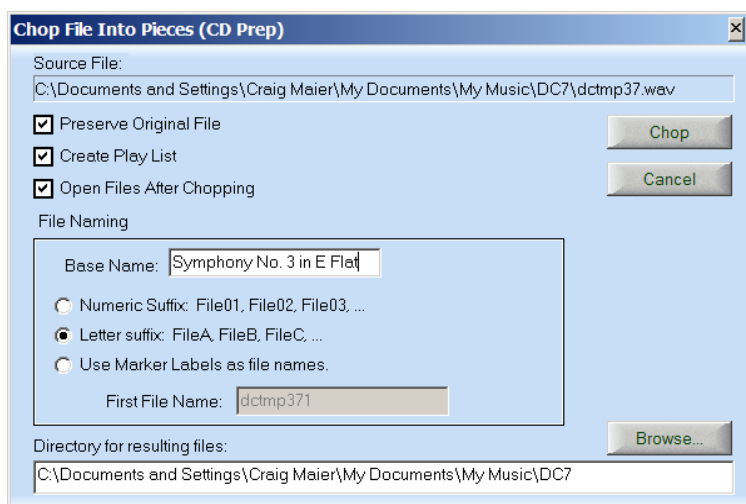


Figure 103 - Chop File Into Pieces

If you've chosen to record a full album length file onto the hard drive and are now ready to break the large file into the smaller "song" files, this feature, used in conjunction with either your markers (manually finding the beginning and ending of your "songs") or with the Find and Mark Silent Passages tool (automatically finding the song tracks by the silence in between two songs...described below) allows you to accomplish this task. It allows you to choose from any or all of the following options:

- **Preserve Original File:** After you've "broken up" your main .wav file, do you still want to keep it as a master?
- **Create Play List:** This launches the Play List editor, which allows you to set up your CD's running order, allows you to add, audition or remove specific titles. The play list also allows you to export your cue sheet for other CD burners in the .cue format. For more info on the playlist, refer to the earlier chapter on the File Menu.
- **Open Files after Chopping:** In most cases, you'll want to audition your "cuts" for accurate beginnings, endings, volume continuity, etc. before actually burning your CD. This checkbox automatically opens your new "cuts" so that can take place.

- **File Naming:** This will list the base name of your Master .wav file and then you can choose whether the resulting “chopped” pieces will be named with letter or number suffixes.
- **Directory for Resulting Files:** Allows you to choose a storage facility for your resulting files.

Find and Mark Silent Passages

DC8/DC FORENSICS includes a feature, which will automatically find and mark the silent passages of your .wav file. This is particularly useful when you desire to process an entire vinyl record album (or tape) in one shot through the various algorithms, and then break them up into separate .wav files at the end of the process. You have the ability to select the threshold of silence, and the time duration of the silence. After you have invoked this feature, you will see all of the markers moved to the silences between cuts. You can move the markers manually, if you are not satisfied with the separations that were automatically determined by the program. After this has been completed, you can chop the file into pieces, and separate .wav files will be created.

Find and Mark Silent Passages will also be found useful for identifying long silent sectors of surveillance recordings automatically. This can save a lot of time for the Forensics examiner who must deal with very long recordings, most of which contains no conversation.

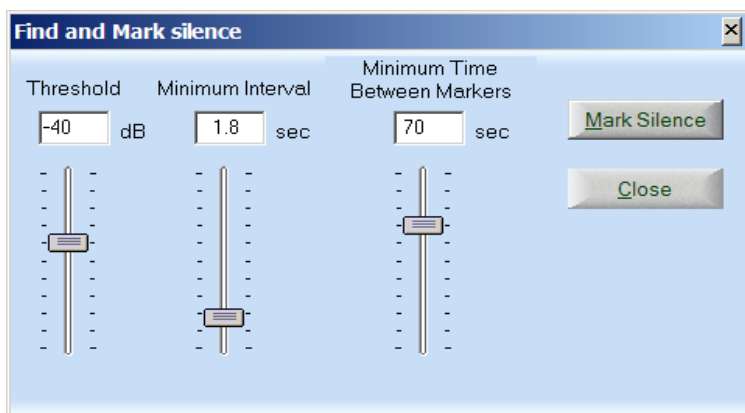


Figure 104 - Find and Mark Silence

Gain Normalize

The Normalize Gain feature searches an entire .wav file looking for the peak signal level. Then, it adjusts the gain applied to the file so that the overall level is below that value. This will provide the best signal to noise ratio and a reasonable volume balance for each "cut" on your final master. Normalize Gain should be applied before burning a CD-ROM or making your final tape.

Important Note:

In Classic Editing mode, Gain Normalize will search the entire file for the peak and adjust the gain based on that peak. It will ignore any selected area. In Fast Edit mode, Gain Normalize will abide by a selected area, find the peak there and adjust the selected area only with the appropriate gain.

Normalized Gain Scaling



It allows you to scale the gain of a .wav file to values other than 0 dB (full-scale output). The range of adjustment provided is +/- 20 dB, which corresponds to a gain factor range of +/-10. If you apply gain scaling above 0 dB, some portion(s) of the .wav file will be clipped. This could be useful in a situation wherein a single transient pulse or two are dominant in the .wav files amplitude, and "clipping" it is irrelevant to you. If you apply scaling below 0 dB, all resultant signals shall be below full-scale output.

Note: Both the Gain Normalize and the Gain Normalize Gain Scaling functions will query you **“Do you want to save UNDO information for this operation?”** If you say yes to this, the process will take about twice the amount of time that’s required if you choose to answer no.

CD Burner



The Diamond Cut CD Burner allows you to create CD’s in conformance with the Red Book Audio CD standard. Its integration into the DC8 software package coupled with drag and drop capability from the DC Tune Database simplifies the CD creation process after a restoration has been completed.

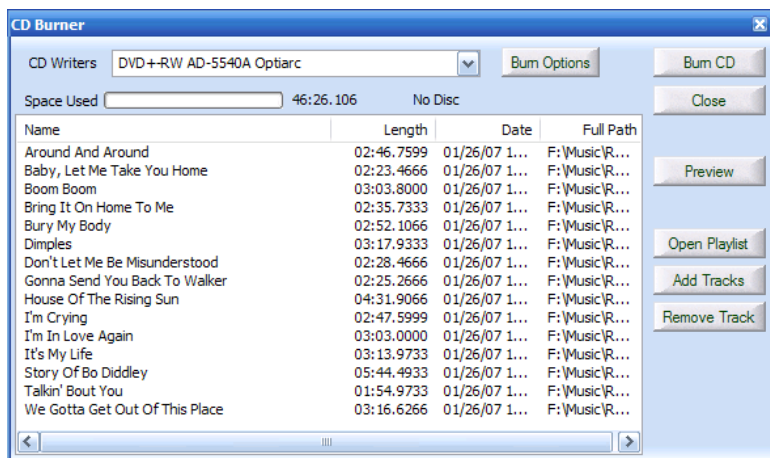


Figure 105 – The Diamond Cut CD Burner

Song lists can be imported directly into the CD burner from your play lists or from the DC Tune Library, and these can be either in the .wav, mp3, or .wma format. The software will automatically convert these file formats for compatibility with the Red Book standard. The software will query your CD ROM burner to determine its maximum speed capability and use that value during the burn process.

To import files from a Play list, click on the “Open Playlist” button and selecting the files that you wish to burn to CD –R. WLS, XML, CUE M3U and PLS playlist formats are supported by the CD Burner. If you have your DC Tune Library open, just drag and drop the desired sound file(s) into the CD burner. You can add or delete tracks from the list using the “Add Tracks” or “Delete Tracks” button(s). For example, to delete a track, just click on it in the CD burner (to highlight it) and then click on “Delete Track”. To hear a particular file in the list before burning the CD, highlight it by clicking on it and then click on “Preview”.

When adding tracks, be sure that the “Space Used” graph does not exceed the capacity of the media that you plan to use, or an error will be generated. Place a blank CD-R into your ROM burner and wait a

few seconds. A message should appear at the top of the Burner routine indicating “Blank CDR. If you are satisfied with your song listing, click on the Burn CD button to commence the CD burning process. A Progress Bar will appear indicating the progress in terms of “% Done”. The system will report when the process has been successfully completed with the following message:

Success: The CD has been burned successfully. Please remove the disc from the drive.

Burn Options

Clicking on the “Burn Options” reveals a number of user selectable alternatives associated with your CD burner. They are as follows:

Maximum Writing Speed: You can limit the burning speed anywhere from 1X to 24X depending on the characteristics of your optical drive. Sometimes, slower burn speeds improve the accuracy of a CD Burn cycle with the obvious tradeoff.

Recording Mode: The system is capable of the two primary CD Burning modes including Disc at Once and Track At Once.

Disc At Once Mode (Gapless) Recording: This feature (DAO) is useful for concert or orchestral recordings wherein one does not want to produce any glitches at the point where the track changes during play. After you drop your markers and before you break your file into pieces, you should Quantize for CD Audio so that this feature works seamlessly.

Write CD Text: Incorporates Text into the file header of your CD when checked. The text information comes from the name field in the file list. It could be the track name if it comes from specific files. Note that only certain CD players are compatible with this feature and can take advantage of this mode of operation.

Track At Once: This mode (TAO) is the most common mode in which CDs are burned. It places a gap between each track. The track is settable anywhere from 0.013 seconds to 20 seconds. The

factory default for this setting is 2 seconds, but it will remember your last data entry value.

Enable Optimal Power Calibration: This optimizes the burn power used to create your CD. This will produce the highest quality and therefore the most robust CD burns. However, it takes a longer period of time to burn a CD when this mode is used.

Note 1: The Diamond Cut CD Burner can import files consisting of the WLS, XML, M3U, and PLS playlist formats.

Note 2: The DC8 CD burner defaults to a 2-second gap between each track selection when Track At Once Mode is used.

Note 3: Some of the CD burner features are only available in the Forensics version of the product.

The View Menu

The View menu allows you to access the commands that affect the manner in which files, controls, and parametric displays are presented to the user. To activate a particular feature, click on it with the left mouse button. Once it has been activated, a check mark will appear next to the chosen command.

The Diamond Cut Spectrum Analyzers

There are two versions of the Diamond Cut Spectrum Analyzer available in the version 7 product family. The DC8 version provides a Standard Precision Spectrum Analyzer providing excellent resolution especially at the low frequency end of the audio spectrum. The DC Forensics version provides a High Precision Spectrum Analyzer with higher frequency resolution over the entire audio frequency spectrum coupled with a display which can zoom in on a specific frequency range via a start and stop frequency feature. The High Precision Spectrum Analyzer also offers the user the option of octave based 10 and 30 band displays for use in acoustical measurement situations.

Spectrum Analyzer – Standard Precision (DC8)

The Spectrum Analyzer command in DC8 brings up the floating Standard Precision Spectrum Analyzer. The Spectrum Analyzer is capable of resolving frequency increments as small as 0.02 Hz on 44.1 KHz sample rate files when it is set appropriately. This Analyzer can be used with any of the filters or effects. It is connected to the output of the filter or effect, so that you can see, in the frequency domain, how you have affected the file. If you want to compare the output of the filter or effect to the input, use the Bypass function on the filter or effect window. Since the Spectrum Analyzer utilizes constant Hertz per frequency band, it will display white noise as a flat, horizontal line. This is unlike octave weighted real time audio analyzers in which white noise produces a diagonal, positive sloped line, and pink noise produces a flat horizontal line. Conversely, pink noise displayed on the Spectrum Analyzer will be displayed as a negative sloped diagonal line. The Spectrum Analyzer displays the algebraic sum of the left and right channels. Keep in mind that the use of the Spectrum Analyzer will slow your system down slightly. Therefore, when you are done using it, shut it down.

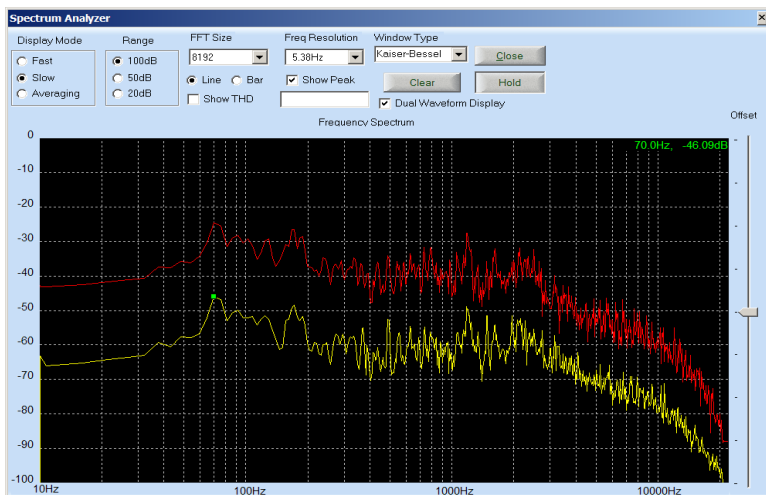


Figure 106 – The Standard Precision Spectrum Analyzer

The following displays are provided on the Spectrum Analyzer:

- **Frequency vs. Amplitude Graph:** The vertical axis indicates 0 dB at the top and ranges down to -100 dB at the bottom. The horizontal axis indicates frequencies from 10 Hz (left) to a little over 20 kHz (right).
- **Two digital readouts indicate the frequency and amplitude of signals feeding the Spectrum Analyzer.** One signal is the peak value and the other is user defined.

The following controls are provided:

- **Display Mode:**
 - **Fast:** Shows the spectrum in almost real time.
 - **Slow:** Shows the spectrum with slower ballistics.
 - **Averaging: On/Off:** This allows the system to provide you with the average signal spectrum rather than in real time display. The averaging interval will be as long as the analyzer is left in operation.
 - **Dual Waveform Display:** This checkbox allows you to view both the left and right channels simultaneously on the Spectrum Analyzer display provided that you are working with a stereo .wav file. The right channel information will be displayed in red while the left channel is displayed in green. If you operate this feature with a monophonic .wav file input, the display will be shown in green.
- **FFT Size:** 11 selection alternatives (64, 128, 256, 512, 1,024, 2,048, 4,096, 8,192, 16,384, 32,768, & 65,536). This selection determines the frequency resolution of the spectrum analyzer. The higher the number selected, the better the frequency resolution of the display.
- **Range:** This scales the vertical axis to 100 dB, 50 dB, or 20 dB full scale. This feature along with the offset control allows you to hone in on a particular signal.

- **Hold Button:** This is a "toggle" function and will allow you to freeze or un-freeze the spectral display update.
- **Clear Button:** This clears the display. It is especially useful when using averaging mode so that you can clear the display after a long averaging interval, allowing you to move onto a new area of the Waveform to be averaged. Clearing the display will re-initiate the averaging process as well as clearing the display.
- **Show Peak Button:** This feature will automatically find the peak amplitude signal and display its Frequency and Relative Amplitude value in the upper right hand corner of the Spectrum Display screen. The marker and display for this feature are red in color.
- **User Controlled Marker:** You can place a marker anywhere you want on the spectral display by clicking the left mouse button on the peak that you are interested in measuring. To accomplish this, merely point the mouse cursor to the peak of interest and click the left mouse button. A green marker will appear at that location and a yellow digital display of the frequency and relative amplitude of the signal that you pointed to will appear in the upper left hand corner of the spectral display. To read another value, merely click the mouse again, pointing to a new spectral line. The marker and the display will then be updated.
- **Frequency Resolution:** The Spectrum Analyzer has the ability to display frequencies with the following values of resolution as indicated on the chart below. Note the interaction between the FFT Size selected and the available Resolution in Hz that are available: "OK" indicates available ranges. "NA" indicates Not Available resolution values by Frequency Band. It is very important to note that small values of FFT Size produce poor values of Resolution, but the response will be very fast. Conversely, large values of FFT Size produce good values of Resolution, but the response will be very slow. This latency time (or display refresh / update) is a function of the Spectrum Analyzer frequency resolution setting. The latency expressed in Seconds, is displayed in the right-hand column for each value of resolution.

Spectrum Analyzer Resolution Chart

FFT Size												Latency
	64	128	256	512	1,024	2,048	4,096	8,192	16,384	32,768	65,536	(In Secs)
Res. Hz												
689.06	OK	NA	NA	NA	NA	NA	NA	NA	NA	NA	NA	.002
344.53	OK	OK	NA	NA	NA	NA	NA	NA	NA	NA	NA	.0039
172.27	OK	OK	OK	NA	NA	NA	NA	NA	NA	NA	NA	.0078
86.13	OK	OK	OK	OK	NA	NA	NA	NA	NA	NA	NA	.0156
43.07	OK	OK	OK	OK	OK	NA	NA	NA	NA	NA	NA	.0312
21.53	OK	OK	OK	OK	OK	OK	NA	NA	NA	NA	NA	.0625
10.77	NA	OK	OK	OK	OK	OK	OK	NA	NA	NA	NA	.125
5.38	NA	NA	OK	OK	OK	OK	OK	OK	NA	NA	NA	.250
2.96	NA	NA	NA	OK	OK	OK	OK	OK	OK	NA	NA	.5
1.35	NA	NA	NA	NA	OK	OK	OK	OK	OK	OK	NA	1.0
0.68	NA	NA	NA	NA	NA	OK	OK	OK	OK	OK	OK	2.0
0.34	NA	NA	NA	NA	NA	NA	OK	OK	OK	OK	OK	4.0
0.17	NA	NA	NA	NA	NA	NA	NA	OK	OK	OK	OK	8.0
0.08	NA	NA	NA	NA	NA	NA	NA	NA	OK	OK	OK	16
0.04	NA	NA	NA	NA	NA	NA	NA	NA	NA	OK	OK	32
0.02	NA	NA	NA	NA	NA	NA	NA	NA	NA	NA	OK	64

Note 1: Because the FFT creates solutions in both the real and imaginary planes, the actual number of frequency bands created is the FFT Size/2.

Note 2: NA = Not Applicable

- **Offset Slider Control:** This allows you to move the centering of the spectral display up or down. It is of particular value when the "Range" control is set to a high sensitivity value such as 50 dB or 20 dB, and the signal appears to be off of the screen. By using the Range control and the "Offset Slider" control, you can zoom-in on a signal of interest.
- **Window Selection:** Seven window selections are provided so that you can establish the appropriate tradeoff between Stop Band attenuation and Lobe Width.
 - A. Blackman
 - B. Hanning
 - C. Hamming
 - D. Rectangular
 - E. Kaiser-Bessel
 - F. Triangular
 - G. Bessel

The following chart shows the performance of several of the provided window functions. Others are provided with the software for completeness.

Window Name	Side-Lobe Peak Amplitude in dB	Main Lobe Band-pass Width	Minimum Stop-Band Attenuation in dB
Blackman	-57	$12 \pi / N$	-74
Hanning	-31	$8 \pi / N$	-44
Hamming	-43	$8 \pi / N$	-53
Rectangular	-13	$4 \pi / N$	-21

Window	Characteristics	Application
--------	-----------------	-------------

Name		
Blackman	Exhibits the best amplitude resolution and accuracy, but has the poorest frequency resolution	Very Good for measuring single frequency signals in order to look at its higher order products.
Hanning	This exhibits fair frequency resolution but poorer amplitude accuracy compared to the Rectangular Window	Good for measuring Narrow Band random Noise, measuring Periodic Signals and for measuring transient bursts wherein the signal levels prior to and just after the event are much different in amplitude.
Hamming	This window is very similar to Hanning, however it exhibits slightly better frequency resolution and accuracy compared to Hanning	Same Applications as Hanning.
Rectangular	Exhibits the best frequency resolution, but has the poorest amplitude resolution and accuracy. In essence, Rectangular is not a window, per se, at all.	Good for measuring sine waves that have similar amplitudes and whose frequencies are loose, for measuring broad band noise with slowly varying frequency spectrum and for measuring Transient bursts wherein the signal levels before and after the key event are similar in amplitude

Note: Hanning is the recommended default window to use for general applications.

- **Display Type:** Choose between Line or Bar (Graph) modes.
- **Display Size:** The overall physical dimensions of the Spectrum Analyzer are user sizable. Simply use your mouse to drag its horizontal and vertical margins to create a display size to your liking.

Distortion Analyzer (THD Mode)

This selector box converts the Spectrum Analyzer into a high performance Distortion Analyzer when enabled. A pure Sine wave obtained from either a very high quality external signal generator or the "Make Waves" generator must be used as the stimulus for the device under test (DUT) or algorithm under test. The THD meter measures the fundamental signal component separating it from the rest of the total harmonic signals presented to the system. It also "notches" out the fundamental and sums the remaining harmonics. Then it takes the ratio of the fundamental to the sum of the measured harmonics and expresses it as a percent on the digital readout. Even and Odd order distortion components are produced by transfer function non-linearity. Even products are dominantly due to transfer asymmetry. Note: The Show Peak checkbox must be checked before enabling the THD display.

Important Note:

When not using the Distortion Analyzer, turn it off to minimize CPU utilization.

- Ultra High resolution Forensics mode: Set the FFT Size to 4,096 or greater. If you then click on 0.67 Hz or smaller, the system will re-scale the horizontal (X) axis of the spectral graph to indicate 700 Hz full scale rather than 20 kHz when you run the spectrum analyzer. You can resolve frequency increments as small as 0.02 Hz as indicated by the settings shown on the resolution chart. To augment this capability, use the "Range" control setting set to 20 dB and the "Offset" control to zoom in on the signals of interest. This combination of features are particularly useful when trying to determine if a forensics recording is a "dub" of the original by looking for two discrete "hum" frequencies on the spectrum. For more information on tape authentication, please refer to the "How do I?" section of this user's manual.

Important Note:

When using the Spectrum Analyzer in Ultra High resolution mode, it will take a long time for the system to integrate. A message will appear at the bottom of the display during this interval stating "Sampling, Please Wait." Refer to the resolution chart for latency details.

Measure Spectrum of Either L or R Channel(s) (Tutorial)

The Spectrum Analyzer displays the algebraic sum of both channels in normal operation. If you want to obtain the spectrum of only one channel, use the following procedure:

1. Bring up the Spectrum Analyzer.
2. Set its parameters appropriate for the measurements that you wish to make.
3. Bring up the File Conversions feature found under the Filter menu.
4. Choose Stereo to Left only or Stereo to Right only, depending on the channel of interest.
5. Click on Preview, and only the chosen channel will be presented to the Spectrum Analyzer.

Measuring the %THD of a component (Tutorial)

The Spectrum Analyzer contains a special feature which allows you to measure the % Total Harmonic Distortion of a piece of electronic equipment, whether it be a CD player or an audio amplifier with a fairly high degree of accuracy. This will require, however, the use of a professional grade, high performance sound card and CD player in order to achieve accurate results. To achieve accurate results, it is necessary that the signal sample rate be at least 44.1 KHz in order to account for all of the Harmonic Distortion products within the audio spectrum. There are two procedures that you can choose from. The first procedure is simple to perform, but less accurate than the second procedure which is more complicated.

Spectrum Analyzer settings for %THD Measurements:

Display Mode: Fast

FFT Size: 16,384

Frequency Resolution: 2.69 Hz

Range: 150 dB

Mode: Bar or Line, your choice

Window Type: Kaiser-Bessel

Show Peak: Check the checkbox

Show THD: Check the checkbox

Note 1: When using the Forensics High Precision Spectrum Analyzer, the Distortion Analyzer feature is accessed via its Options Menu.

Note 2: If you are using the High Precision Spectrum Analyzer found in the Forensics version of the software, set the Start Frequency at 10 Hz and the Stop Frequency at 22,050 Hz.

▪ **Procedure #1 (Simple Method):**

1. Create a 5 minute stereo 44.1 kHz sampled 1000 Hz* Sine waveform .wav file by using the Make Waves feature.
2. Name the file, and make sure you know the path on which it was stored.
3. Use your ROM burner software to make a CD of this .wav file.
4. Connect the output of the CD player to the component under test's input
5. Connect the output of the device under test (DUT) to your sound card's line input.‡
6. Launch DC8/DC FORENSICS
7. From the View menu, enable the Spectrum Analyzer.
8. In the Spectrum Analyzer, enable the THD meter feature.
9. While operating in "Live" mode, play the CD.
10. You should see a dominant spectral spike at 1 kHz.
11. The Distortion Analyzer will calculate the %THD and display it in the THD meter window.
12. Done

▪ **Procedure #2 (High Accuracy Method):**

1. Perform steps 1-3 from the above tutorial
2. Connect the output of the CD player to the line input of your sound card.
6. Launch DC8/DC FORENSICS.
7. From the View menu, enable the Spectrum Analyzer.
8. In the Spectrum Analyzer, enable the THD meter feature.
9. While operating in "Live" mode, play the CD.
10. You should see a dominant spectral spike at 1 kHz.

11. The Distortion Analyzer will calculate the CD and sound card %THD and display it in the THD meter window. Write this value down and call it %THD(I) (instrumentation).
12. Disconnect the CD player from the line input of the sound card and then plug it into the electronic device under test's (DUT) line input.
13. Connect the output of the DUT to your sound cards line input.‡
14. In the Spectrum Analyzer, observe the THD meter feature and write down the %THD indicated. Write this value down and call it %THD(T) (total).
15. Calculate the DUT distortion using the following equation:

$$\%THD(DUT) = (((\%THD(T))^2) - ((\%THD(I))^2))^{1/2}$$

Note: Lower Frequency .wav files can be used.

‡ **Note:** If a power amplifier is being tested, appropriate resistive loading must be connected to its output, along with an appropriate attenuator between the amplifiers output and the sound card input. DO NOT connect the output of a high-output audio power amplifier directly to the input of your sound card. Doing so could result in the destruction of your sound card.

Spectrum Analyzer – High Precision (DC Forensics)

A High Precision Spectrum Analyzer is provided in the DC Forensics version of the Diamond Cut version 8 product family which is sometimes refer to as an FFT analyzer. It uses techniques including higher FFT sizes, chirp Z-Transforms, decimation methodology, a widened array of window types, waveform storage, and a new graphical display interface in order to achieve its higher level of precision compared to its standard precision counterpart. This high precision spectrum analyzer is useful for making acoustical measurements, authenticating forensics audio tapes by detecting dubs, verifying that forensics tapes have not been edited, assuring compliance with OSHA Noise Standards (both broadband as well as pure tones), performing mechanical vibration testing, measuring control loop performance, and many other applications.

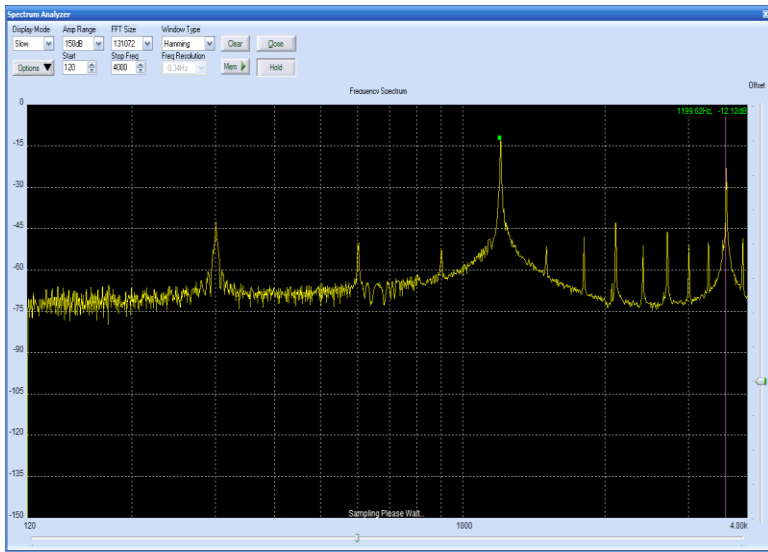


Figure 107 – The High Precision Spectrum Analyzer

The High Precision spectrum analyzer is described here highlighting the features which are different compared to that of the Standard Precision version found in DC8. It is highly recommended that you read the section that precedes this section before learning about the High Precision Analyzer, since many features are shared by both. Here are the salient differences provided by the High Precision Spectrum Analyzer:

- Added Zoom capability with a Start and Stop setting coupled with Frequency Axis Scrolling Capability
- Added a Memory feature to store and recall spectrums, both Left + Right and individual Left and Right Spectrums
- Added a Vertical Marker Measurement System
- Added Hi-Resolution mode so that all of the FFT bins are applied to the frequency span selected by the user. For example, if you have an FFT size of 4096, and a 1 octave

span, then all of the 4096 bins would be spread over just that single octave presenting the user with a very high frequency resolution display.*

- Added Kaiser 10, Kaiser 20, and Welsh Window types
- Increase FFT Size to 132K maximum
- Added 1.0 Octave and 1/3 Octave Analog Filter based modes for noise and acoustical measurement applications.

*Note: The selectivity of the High Precision Spectrum Analyzer is still limited by the basic FFT size and the Window choice.

The first difference that you may note between the DC8 and the DC Forensics High Precision Spectrum Analyzer is the existence of a drop-down menu on the left side of the system labeled “Options”. Clicking on this menu provides you with the following options from which to choose.

Option Menu

- Show Peak Checkbox – Provides for the automatic display of the Peak Amplitude Signal level in dB and its Frequency. The two numeric values are shown in Green in the upper right hand corner of the Spectrum Analyzer. Its format is: Frequency in Hz, Level in dB (Hz, dB).
- Dual Waveform Display Checkbox – When this checkbox is not ticked off, the system displays the summed average of both the Left and Right channels. When it is checked, it provides you with the ability to discretely display the Left and Right channels. The Right channel information will be displayed with a Red trace while the Left channel is displayed in Green.
- Bar Graph Mode Checkbox – When this checkbox is unchecked, the system displays each data point with a

line drawn in between. When it is checked, the data is represented in a bar graph format.

- Standard FFT Checkbox – This applies the number of bins (FFT Size / 2) to be distributed over the entire audio spectrum.
- Hi Precision FFT Checkbox – This mode allows the number of frequency bins to be applied strictly to the range (or span) of frequencies selected by the user. This mode effectively improves the Frequency resolution of the system when the range of frequencies selected is less than the entire audio spectrum.
- 10 Band Octave Analyzer Checkbox – This changes the mode of operation from an FFT Based system to an IIR based simulation of an Analog, Octave frequency weighted system.
- 30 Band Octave Analyzer Checkbox – This changes the mode of operation to an IIR based simulation of a 1/3rd Octave frequency weighted system.
- Smooth Spectrum – This smoothes the graphical display based on a set of frequency domain calculations using adjacent sides of a bin within a fft.

Note: The two IIR based Analyzers are useful for measuring relative noise levels against local Environmental laws, especially pure tones. For more information about pure tones, please refer to the Glossary of Terms section of this Users Manual.

Note 1: The 10 Band Analyzer breaks the audio spectrum down into the following center frequency buckets:

31 Hz, 62 Hz, 125 Hz, 250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz

8 kHz, 16 kHz

Note 2: The 30 Band Analyzer breaks the audio spectrum down into the following center frequency buckets:

25 Hz, 31 Hz, 40 Hz, 50 Hz, 62 Hz, 80 Hz, 100 Hz, 125 Hz, 160 Hz,
200 Hz, 250 Hz, 320 Hz, 400 Hz, 500 Hz, 640 Hz, 800 Hz, 1 kHz,
1.3 kHz, 1.6 kHz, 2 kHz, 2.5 kHz, 3.1 kHz, 4 kHz, 5 kHz,
6.2 kHz, 8 kHz, 10 kHz, 13 kHz, 16 kHz, 20 kHz

Display Mode

- **Fast:** Shows the spectrum in almost real time with a rapid sample rate.
- **Slow:** Shows the spectrum with slower / averaging ballistics.
- **Averaging (On/Off):** This allows the system to provide you with the average signal spectrum rather than in real time. It will integrate for as long as the file is being previewed, played, or loop played. Its ability to respond to changes becomes slower as the integration interval lengthens. In other words, the longer that the file is played in this mode the slower its response will be and the more averaged will be its resultant display.

Amp Range (Amplitude Range)

This feature controls the Vertical Axis full scale range. You can select between the following:

- 150 dB
- 100 dB
- 50 dB
- 20 dB

Note: Lower values of dB produce greater degrees of amplitude resolution.

FFT Size

The FFT Size determines the basic frequency resolution of the system with the number of frequency bins equal to the FFT size / 2. Higher FFT sizes produce higher selectivity or frequency resolution with the

tradeoff of longer integration time. You can choose between the following FFT sizes:

64, 128, 256, 512, 1024, 2048, 4096, 8192, 16384, 37768, 65536,
131072

Start and Stop Frequency Values

The Start Frequency and Stop Frequency Data Entry fields allow you to establish the horizontal axis frequency range displayed on the Spectrum Analyzer. The Start Frequency sets the left-most graphical setting while the Stop Frequency sets the right-most setting. The frequency set-ability of this system is 1 Hz for each of the two parameters. The maximum range of adjustability for Start and Stop Frequency runs from 1 Hz to half the file sample rate*. So, by way of example, using a 44.1 kHz file, you can set a Start Frequency as low as 1 Hz and a Stop Frequency as high as 22,050 Hz (which covers more than the entire audio spectrum of 20 Hz to 20,000 Hz). On the other hand, you can set a frequency range as small as 1 Hz. For example, you can set a Start Frequency of 59 Hz and a Stop Frequency of 60 Hz. It is important to note that sample theorem comes into play and the highest frequency that you can actually plot with the Spectrum Analyzer shall be the .wav file sample rate / 2.

*Note: As an example, a 44.1 kHz sampled rate file will allow for a maximum Stop Frequency of 22,050 Hz. Similarly a 96 kHz sampled file will allow for a maximum Stop Frequency of 48,000 Hz.

Frequency Axis Scroll Bar

At the bottom of the Spectrum Analyzer display, you will find a scroll bar. This allows you to move around in the spectrum display using your left mouse button coupled with a drag motion, either left or right in direction.

Window Type

You can choose between the following Window types:

Bessel, Blackman, Hanning, Hamming, Kaiser 10, Kaiser 20, Kaiser-Bessel, Rectangular, Triangular, Welsh

Descriptions of the characteristics of many of these Window types can be found in the section describing the Standard Precision (DC8) Spectrum Analyzer.

Mem (Memory) Button

This button activates the tracer storage and trace recall features of the High Precision Spectrum Analyzer. You can store a large number of Spectrum Analyzer traces in the directory of your choice. The file extension is .spt for these data. You can only display one set of recalled data at any given time along with the present display data. This feature is very useful for comparing spectral data to a reference set of data and presenting it on the same graphical display. Clicking on it (Mem) brings up the following options:

- Save Trace: Browse to the Directory of your choice.
- Save Left Trace: Same as above, but applies to the Left Trace only.
- Save Right Trace: Same as above, but applies to the right Trace only.
- Load Trace: Browse to the directory containing the trace to be recalled and click on it and it will appear in the Spectrum Analyzer Display.
- Remove Trace: Eliminates a Memory based displayed signal trace. Clicking on this button clears the Spectrum Analyzer Display Window of the previously recalled spectral data.

Close Button

Clicking on this button closes down the Spectrum Analyzer Display Window.

Hold / Run Button

Clicking on this button alternately freezes (holds) the display or runs the Analyzer in a toggle manner of operation.

Point, Click and Measure

To measure the Amplitude and Frequency of any given spectral line or signal, just point your mouse at the signal of interest and left – mouse click. The two numeric values are then displayed in Yellow in the upper left hand corner of the Spectrum Analyzer. Its format is: Frequency in Hz, Level in dB (Hz, dB).

Settings for High Accuracy Measurement of %THD with the Precision Spectrum Analyzer

Display Mode: Fast

FFT Size: 65,536

Start Frequency: 20 Hz

Stop Frequency: 20,000 Hz

Frequency Resolution: 0.67 Hz

Resolution Mode: Check “Standard FFT” in the Options Menu

Range: 150 dB

Display Mode: Bar or Line, your choice as found in the Options Menu

Window Type: Kaiser-Bessel

Show Peak: Check this checkbox in the Options Menu

Show THD: Check this checkbox in the Options Menu

Using the Spectrum Analyzer in Real Time Mode

1. Bring up the Precision Spectrum Analyzer and set it up as desired.
2. Bring up the Multifilter and remove all of the filters from its lineup or click on the “Bypass” checkbox.
3. Click on “Live Preview”
4. The Spectrum Analyzer will then display the frequency domain content of your signal in real time as presented to the input of your soundcard after a relatively short latency time period.

XY Display

The X-Y Vector Display displays the phase relationship between two waveforms. The instantaneous vector is displayed with a green trace while the averaged angle is displayed in red (provided that the

averaging mode is enabled). **The Averaging feature is enabled by using the Averaging Selector and choosing between “None”, “Short”, or “Long”. “None” turns off the averaging display leaving only the instantaneous trace active on the XY display. “Short” displays the moving averaged vector displacement angle between the two signals over a 4095 samples interval. “Long” displays the average vector displacement angle over the interval commencing at the time when the play button is started until the time when the stop button is depressed or over the interval of time highlighted in the play source display. The averaging interval for the “Long” mode is therefore a variable determined by the user depending on the length of the played file or the highlighted area of the waveform selected to be played and therefore measured. The Averaging mode is very useful when trying to measure the phase angle of signals having significant amounts of phase jitter. (Low cost tape recorders often display high levels of phase jitter especially at the upper frequencies in the audio band making the instantaneous graph difficult to use while making azimuth adjustments to the units playback head.)

This feature is primarily used to align the azimuth of analog tape deck recording and playback heads. To do so, a known pre-recorded azimuth alignment tape is required containing a fixed frequency tone. The X-Y Vector display in conjunction with your azimuth tape and the Time Offset feature can be used to fine-tune a recorder's time alignment without the necessity of having to take a screwdriver to your tape deck's head alignment adjustment screws. The goal here is to correct the azimuth of the .wav file using the File Conversions Filter with the Time Offset slider control. Preview the .wav file with the X-Y Vector Display showing, and adjust the Time Offset slider control until a 45 degree positive sloped (up and to the right) straight line is seen on the display. That will be the optimum value of azimuth correction.

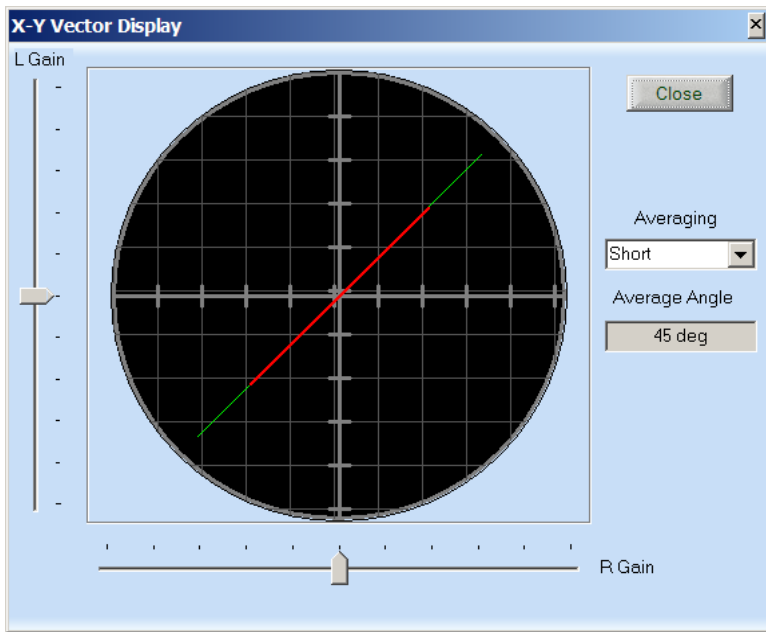


Figure 108 - X-Y Vector Display Showing Proper Azimuth Alignment

The following features will be found on the X-Y Vector display:

- X-Axis displacement (Horizontal), which corresponds to the Left Channel Input
- Y-Axis displacement (Vertical), which corresponds to the Right Channel Input
- X-Axis gain control slider (Horizontal in position)
- Y-Axis gain control slider (Vertical in position)
- Averaging Selector**
 - None: Averaging function Turned Off
 - Short: Averaging Interval = 4095 Samples
 - Long: Averaging Interval is Variable depending on length of “played” file
- Average Angle Display Window**: Displays the Average Angle value with up to 6 digits of resolution.

Here is a listing of some vector displacements, which can be observed on the X-Y Vector Display. They have the following meaning and are sometimes referred to as Lissajous figures:

- Straight line at 45 degrees with a positive (up and to the right) slope = Signals are in phase and are Monophonic.
- Straight line at 45 degrees with a negative (down and to the left) slope = Signals are out of phase and are Monophonic. An example of a situation that could cause this would be a miss-wired Stereo Phono cartridge.
- Straight horizontal line only = Monophonic Right channel only signal.
- Straight vertical line only = Monophonic Left channel only
- Straight line at 45 degrees with a negative (down and to the left) slope = Signals are 180 degrees out-of- phase
- Circle = Signals are 90 degrees phase shifted
- Frozen vertical figure "8" = Signals are frequency phased locked to one another but 2:1 in frequency ratio.*
- Moving vertical figure "8" = Signals are not frequency locked, but are about 2:1 in frequency ratio.*
- Random pattern of squiggly lines on the screen = Stereophonic audio signal

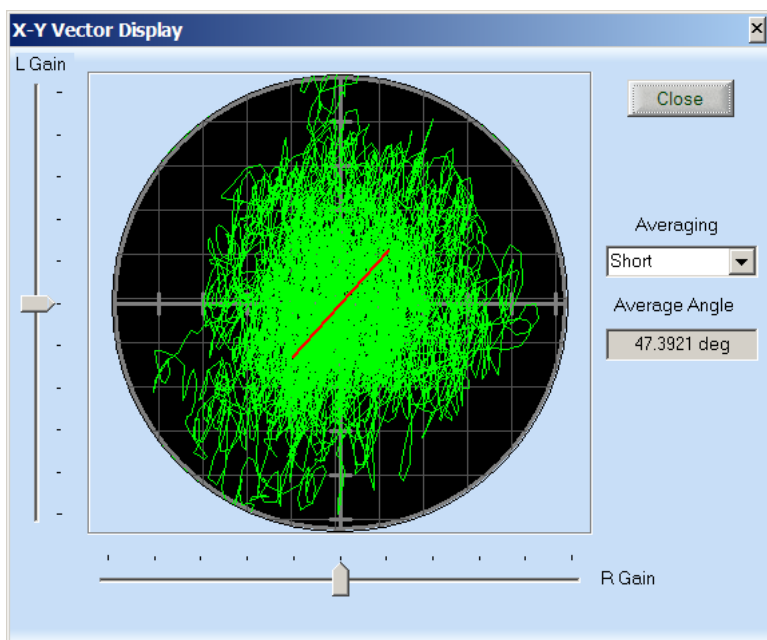


Figure 109 - X-Y Vector Display Showing A Stereophonic Signal

***Note 1:** In these examples, the right channel would be twice the frequency of the left input. If the figure 8 were lying on its side, then the left channel would be twice the frequency of the right input.

****Note 2:** The Vector Angle Averaging feature is only available on the DC Forensics version of the software.

Time Display

The Time Display view displays all of the time related parameters associated with a source or destination workspace. When activated, a display window will appear containing four sets of timing numbers. This window can be dragged and placed anywhere in your workspace. The following time related parameters are displayed:

1. Cursor Location (largest numerals).
2. Start Location of a highlighted area (small numerals).
3. Stop Location of a highlighted area (small numerals).

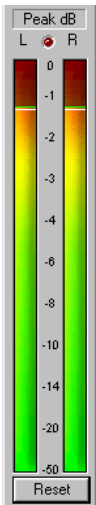
4. Span - This represents the total time of a highlighted area (small numerals).



Figure 110 - Time Display Window

You can monitor “Samples” rather than “Time” in this display if desired. Go to the Preferences Menu and choose the “Display” option. Under that category, you will find a “Display Time Format” choice. Click on “Samples”.

Output VU Meter



Two 100 segment VU meters can be displayed which will indicate the output of any filter or multiple filters that are being used by DC FORENSICS or DC8. These meters indicate the level of the left and right channels and have both average and peak reading ballistics. They are calibrated to indicated values from -50 dB to 0 dB, with 0 dB being full-scale output. Any signal above that level will be clipped by the system. Signals that are clipped are indicated by the illumination of a small red “LED” indicator at the top and in the middle of the meter display.

Peak Hold

The meter includes two peak indicators. The white horizontal bar is designed to “hold” the indication of an overload for two seconds after it occurs so that it will be more obvious to you that a clipping event has occurred. The green horizontal bar is a peak hold indicator and will hold the value of the audio peak until such time as the reset button, located at the bottom of the VU meters, is activated by the mouse. These VU meters can be activated or de-activated under the View Menu. Also, they can be “dragged and dropped” anywhere on your desktop workspace using

your left mouse button or docked and enlarged at the right hand margin of the software user interface.

Volume Control

This feature allows you to view and adjust the sound card primary volume controls. Two controls are presented:

1. Main Volume Control
2. Wave Volume Control

Important Note:

This volume control is only useful for sound cards that make use of the Windows Mixer. For any card that doesn't use the Windows Mixer, adjustment of volume will take place using the proprietary mixer of that product.

Fast Edit History

When working in Fast Edit mode, this cool little window is like having your own personal secretary. It keeps track of every edit and allows you to navigate quickly from one to the other. If you decide to start at midpoint in this list and want to eliminate all edits done after this point, just double click on that edit and the system will ask if you're sure...then eliminate all edits after this point. All of the data pertaining to the fast edit history profile is contained in .ses files. These files are stored with your .pkfs and can be deleted once you've finished with the file. Fast Edit temp files are held in the same directory as your source file and include elements of the source file name for ease of identification.

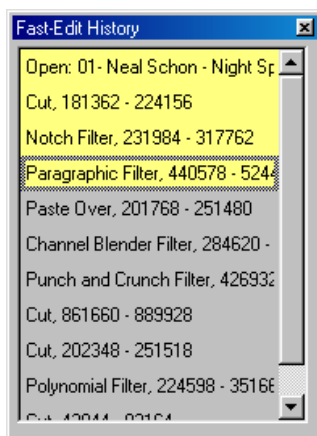
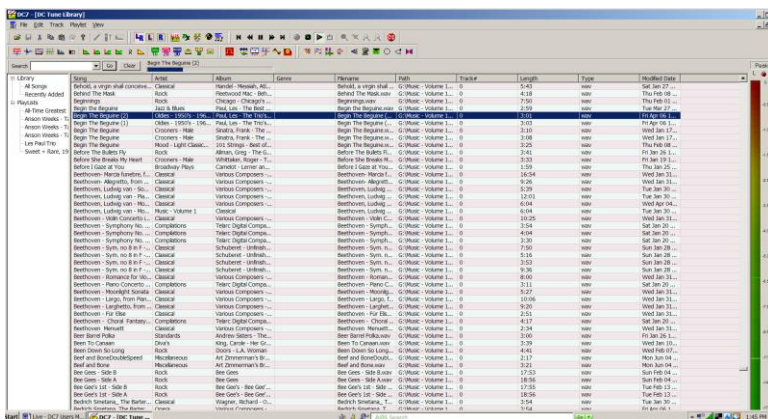


Figure 111 - Fast-Edit History View

DC Tune Library

The DC Tune Library provides you with a full-featured audio file archive capability within the context of your DC8 software. It will search your hard drive and find all audio files and store their path, name, genre and other pertinent data within its file structure. After this database is constructed, it provides lightening speed access to any of your audio files, whether they are in .wav, mp3, or wma format(s). You can search your database, order it in terms of various parameters and create and recall playlists. You can play any file directly through the normal play features of DC8, or you can listen to a file or series of files by using the Preview feature found on the Diamond Cut filters and/or effects. You can even listen to the database by way of a complex series of filters and effects by previewing a file via the DC8 Multifilter.

Details pertaining to the various functions associated with the DC Tune Library can be found in an earlier section of this users guide dedicated to this topic.



Zoom In



Zoom In allows you to magnify any portion (no matter how small) of your .wav file. It can be used with either the Source, Destination or Fast Edit workspace. The Zoom-In process may be repeated any number of times for a really close and detailed look of your audio waveforms. However, only the last 5 zoom levels are remembered. You can also access this feature by using the **Z** hotkey or the Right Click Menu. Your mouse wheel will also control the zooming In - Out feature.

Zoom-In X2



Allows you to Zoom in on a waveform by a factor of 2. The amount of time displayed is decreased by a factor of 2 each time the Zoom-In button is pressed. This zoom differs from the other Zoom function, which zooms in on the selected area. Pressing the + key on the numeric keyboard will also activate the zoom function. It can be used with either the Source or Destination or Fast Edit workspace. The Zoom –In process may be repeated any number of times for a really close and detailed look at your audio waveforms.

Zooming-In/Out on portion of a Wave file (Tutorial)

To Zoom-In and Zoom-Out on a portion of a .wav file, use the following procedure:

1. With your mouse, left click on the workspace location that you desire to magnify.
2. As you begin to drag the mouse pointer towards the right-hand portion of the screen, you will see two black timing reference markers appear in the workspace.
3. The first marker will remain at the location at which you began to perform the "mouse-drag" operation, indicating the location of the "start" position of your zoom region of the .wav file.
4. When you get to the end of the interval of the .wav file portion you wish to Zoom-In on, release the left mouse button, and the second line will remain at that location. This line will have indicated the location of the "end" position of your selected zoom-in region of the .wav file.
5. Click the mouse on the Zoom-In icon
6. The selected area will be re-displayed with the time axis expanded to fill the monitor screen.
7. To zoom back out, merely click with the left mouse button on the Zoom-Out button on the toolbar.
8. After you have clicked on the Zoom-In icon, you will notice that the Time Axis Slider control at the bottom of the file will move to the relative position of the .wav file where you began your zoom operation. By using either the slider directly with your mouse, or by using the arrows at each end of the slider, you will be able to move through the original relative position of the entire .wav file, displaying the same level of time magnification that you had just established through the use of your Zoom-In control.

The Zoom In X2 and Zoom Out X2 buttons function in a different manner and give you the option of zooming in or out without changing the selected area of the .wav file. Pressing Zoom-In X2 will take whatever area of the waveform that is currently displayed and magnify it by a factor of 2. Similarly, the Zoom-Out X2 button will take the displayed portion of the .wav file and decrease the time resolution by a factor of 2. Both of these zooming operations can be performed multiple times if a higher degree of focus or de-focus is desired.

Zoom Out



This key performs the inverse function of the Zoom-In key. It allows you to progressively back out of a .wav file which you had previously Zoomed-In on. Hitting this key allows you to *backpedal* through your last 5 zooms. You can also access this feature by using the **X** hotkey or by using the Right Click Menu. Your mouse wheel will also control the zooming In - Out feature. For a brief tutorial, see above.

Zoom-Out X2



This feature allows you to back out of the displayed area by a factor of 2. The amount of time displayed is increased by a factor of 2 each time the Zoom Out button is pressed. Pressing the – key on the numeric keyboard will also activate the Zoom function. It can be used with the Source, Destination or Fast Edit workspace.

Zoom Out Full

This feature allows you to quickly zoom back to square one. It will display your full waveform on the screen. You can also access this feature by using the **Ctrl + Z** hotkey or the Right Click Menu.

Zoom to Markers

This feature is only activated when markers are dropped. When used, it fills the screen with the material located between your two markers.

Box Zooming

To zoom into an area of a waveform without time or amplitude limits, you can use Box Zooming. This is accomplished by holding down the Shift key and then holding down the left mouse button to draw a rectangle around the area you want to zoom in on. When you release the mouse button, the system will zoom into the circumscribed area of the file. To zoom back out, just use one of the zoom out commands.

Sync Files

This feature, when enabled, mutually synchronizes the Filter and other commands such as Zoom-In and Zoom-Out between the Source and the Destination .wav files. This feature is useful for selective filtering or viewing of a portion of a .wav file. For example, if you run a particular filter, and then find a sector (or sectors) that need further filtering, you can highlight only the portion that needs the additional processing, and apply the filter accordingly. Sync mode will also accommodate different filters being applied to different sectors. Also, when enabled, highlighting and Zooming-In on a particular area of a source file will correspondingly Zoom-In on the same timing co-ordinates in the Destination file. The converse is also true such that Zooming-In on particular set of co-ordinates in a Destination file will also cause the Source file to contain the same Zoom-In co-ordinates. This feature is of particular use when you want to visualize how a particular filter may have modified a waveform from your Source file. Clicking on it with your mouse enables this feature. When it is active, a check mark will appear to the left of it.

Note: The Sync Files function is only useful in Classic Editing Mode.

To use Sync Mode to apply selective filtering to a .wav file, it is first necessary to create a Destination file from your Source file. This can be done through the normal processing procedure of any of the filters, or it can be accomplished with one of the file conversion options such as Mono-to-Mono or Stereo-to-Stereo. Thereafter, merely highlight the portion of the Source file that needs filtering, click on the appropriate filter, select the appropriate filter values and run the filter. Only the highlighted sector of the two files will be enacted upon by the filter, with the results appearing in the Destination file. If another filter (or the same filter that you had just been working with) is then required to be applied to another portion of the .wav file, just repeat the outlined process.

Non-Sync mode of operation

In non-sync mode, the highlighted section of the source file is read and processed by the filter. The processed section is then written to the destination file starting at the beginning of the file. If a destination file already exists, it will be overwritten (a prompt warns you of this). This

mode is useful when only a section of the source file needs to be extracted, or for testing a filter's settings before processing an entire file.

Selective Filtering with Sync Mode (Tutorial)

1. Open the desired Source .wav file.
2. From the Source .wav file, create a Destination .wav file utilizing one of the File Conversion alternatives or one of the DC8/DC FORENSICS Filters.
3. Activate Sync Mode by clicking on it in the View Menu. A check mark will show that it's active.
4. Highlight the portion of your Source .wav file that needs Selective Filtering.
5. Left click on the "Filter Menu."
6. Left click on the filter that you desire to run on the selected portion of your Source .wav file.
7. Adjust the Filter parameters accordingly utilizing "Preview Mode" if necessary to get the parameters to sound appropriate to your taste.
8. Click on "Run Filter."
9. The highlighted portion of the Source .wav file will be processed and inserted into the same time-slot in the Destination Workspace.
10. If further selective filtering is required on a different portion of the Source .wav file, repeat steps 4 through 9 until you are satisfied.

Skins and Themes

Skins and Themes - - - have it your way; this feature offers different skins and themes for your Diamond Cut windows, filters, effects and dialog boxes.

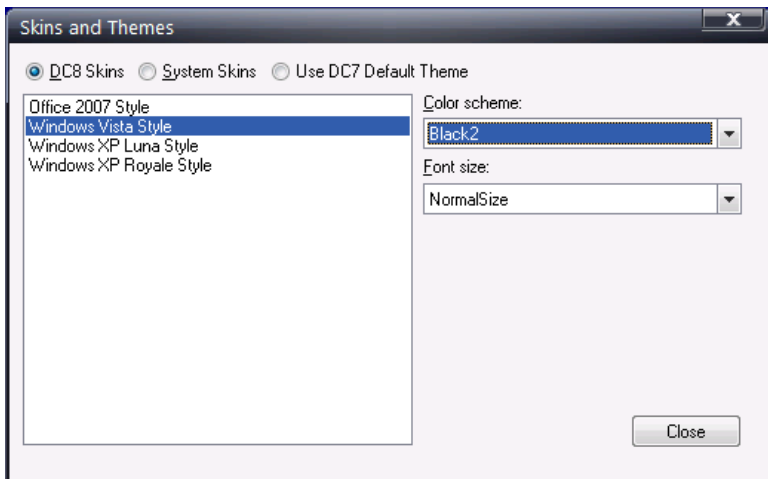


Figure 113 – The “Skins and Themes” Dialog Box

You can choose between a wide variety of skins, themes, colors and fonts. Experiment with this until you obtain your favorite look, or if you get bored with a given look, change it on occasion for variety.

If you prefer the old way that DC7 looked, you can get back to that appearance by choosing the “Use DC7 Default Theme” selection. Also, please note that not all skins and themes support variable color schemes or font sizes. The “Display” preference for showing the dialog color schemes (found under the Edit\Preferences\Display tab) is by default left on when you enable a theme. Users may want to turn these off to see the full effect of the display theme that they have chosen.

Enabling or Disabling ToolbarsInDC8/DC FORENSICS

The following toolbars may be turned on or off by the user by simply checking or un-checking them from the View menu item.

Channel Toolbar

Just beneath the filter and effects toolbar and towards the left of the display can be found the Channel Selector Toolbar. This allows you to

choose which channels are to be operated on by the system. Your choices are as follows:

1. L/R (Both channels)
2. L (Left channel)
3. R (Right channel)

The Channel Selector applies to all Filters, Effects and Editing functions except for tools that change the overall length of the file. Generally, one would tend to leave this feature in the L/R (Both channel) position unless dealing with special editing or sound restoration issues.

Note: You can click on any of the Channel controls while playing to select either the Left, Right or Both channels. When previewing a filter, both channels will always be previewed, but only the channel selected before the preview starts will be affected by the filter.

File Toolbar

The main DC8/DC FORENSICS File Toolbar resides along the top of the program window. It contains the most commonly used file functions of the program. Clicking on them with the left mouse button activates them.

Filter and Effects Toolbar

The Filter and Effects Toolbar also resides near the top of the program window. It contains buttons for the most commonly used filters and effects. This toolbar, like the others, "floats" and can be dragged and dropped anywhere within the DC8/DC FORENSICS window using the mouse. Clicking on them with the left mouse button activates them.

Forensics Toolbar

The Forensics Toolbar contains buttons for the most commonly used Forensics filters.

Play Controls Toolbar

This toolbar contains the “Transport” controls for recording and playback of the audio.

Status Bar

The Status Bar resides along the bottom of the program window. Four parameters are displayed therein:

- **Program Mode** is displayed in the left-hand field. Initially, it will indicate Ready. Clicking on the various toolbar buttons (from left to right) will enable the following functions and indicate the following operation on the status bar:
 - A. Open an existing Waveform audio file
 - B. Save the active Document
 - C. Delete a Section of a .wav file
 - D. Copy selection and put it on the clipboard
 - E. Paste Clipboard contents over the selected area
 - F. Context Sensitive Help
 - G. Rewind file position to beginning
 - H. Pause Playback of the selected file
 - I. Fast Forward to the end of the file
 - J. Start Recording an Audio file
 - K. Stop Playback of the Current file
 - L. Play the selected File
 - M. Zoom-In to the selected area of the sound file
 - N. Zoom-Out to the last selected sound file

Note: Clicking on the various menu items will also activate a one-line help text file, which describes each item.

- **Channels Used:** The field to the right of the "Program Mode" field shows the number of channels being used and will indicate either mono or stereo.
- **Sample Rate:** The next field moving towards the right shows the sample rate in kHz.
- **Bit Rate:** The next field moving towards the right shows the bits per sample of the current file.

- **Highlighted Length:** The next field tells you the length of time for a highlighted portion of the Source or Destination workspace.
- **Available Hard Drive Space:** The next field indicates the Available Hard Drive Space in Mbytes. This is useful to determine whether or not there is enough storage space to perform your processing.

File Info

The following information will be displayed regarding the current highlighted .wav file when "File Information" is clicked:

1. File Name
2. File Path
3. File Size (Bytes)
4. Length (Time)
5. Channels (Mono/Stereo, etc.)
6. Sample Rate (kHz)
7. Bits Per Sample
8. Last Modified (Date)

Rebuild Peak File

Peak files (.pkf) provide a graphical representation (1 sample per 200 and extracting the peak value in that window) of the .wav files and are stored along with any .wav file created or operated upon by DC8/DC FORENSICS. After many editing processes, it may be desirable to update this file to more accurately represent the present state of the displayed .wav file in the source or destination workspace. To do this, click on this function, and you will see that the displayed .wav file is updated, accurately representing your edits. Peak files may be deleted by the user at any time.

The Window Menu

The Window Menu contains all of the commands that affect the manner in which the files are displayed to the user. The following commands are available:

Cascade

The "Cascade" command arranges all of the open Windows so that they appear on the display one behind the other. The most recently opened Window will appear on top, with the remainder arranged in the order in which they were opened.

Tile

The "Tile" command arranges all of the open Windows so that they appear on the display in a matrix configuration (in the same plane) on your display. This feature is used when you need to see all of the workspaces concurrently.

Arrange Icons

The "Arrange Icons" command will arrange all of the "iconized" windows in a row near the bottom of the workspace.

Close All

As the name implies...click here to close all windows and files. The system will verify your choice, ask if you want to save and then close each window and file.

Open Files

A listing of all of the open files is provided at the bottom of the Window Menu. The file that is active will have a check mark indicated to the left of the file name.

The Help Menu

With this product, we strive to provide you with the best possible help. To that end, in addition to this in-depth manual, we also include a computerized help file that is always available at the click of a mouse.

Tip of the Day

This feature is normally displayed on the splash screen. It can also be invoked under this menu item. It provides useful hints on the operation of the software contained within roughly 250 tips and advances through the list one item at a time each time the software is opened. When it gets to the last tip, it returns to the first item in the listing and starts the sequence over again. If you are not interested in using this feature, uncheck the box that indicates "Show Tips on Start-Up."

Restoring a Recording

This feature links you to a section of the help file that describes step-by-step details regarding the restoration of an old recording.

Restoring the demo1.wav

This feature links you to a section of the help file that describes in step-by-step detail the process for restoring the demo1.wav file that is included with the software package. This is an excellent exercise to go through to get an initial idea of the operation of the system.

Contents

The DC8/DC FORENSICS program contains an extensive Help file. These topics are printable from within the help file for your convenience. The following types of information are available:

- DC8/DC FORENSICS program Overview
- Tutorial Information on each filter
- Step-by-Step procedures for utilizing each filter and editing feature
- Audio Restoration process procedures
- Practical Hints on recording procedures
- Extensive Glossary with definitions, tables of values, and appendix information.
- Licensing Agreement, Warranty, and program History information

Context Sensitive Help



This button will provide you with on-line context sensitive Help from the DC8/DC FORENSICS Help file.

Using Context Sensitive Help (Tutorial)

Method #1

1. Point the mouse pointer to the portion of the DC8/DC FORENSICS program display about which you would like some Help file information.
2. Depress the special function key F1.
3. Help file information will appear in a window. This information is usually only summary procedural information on the particular topic. For more detailed information, refer to the actual Help file, and look for tutorial information on the topic of interest.
4. When done, click on the - button at the top left-hand side of the Help window.

Method #2

1. Using the left mouse button, click on the button with the “?” contained within its perimeter on the toolbar.
2. A black, mouse controlled “?” will appear next to the mouse pointer.
3. Move the pointer to the area of the program display about which you would like some Help file information.
4. Click on the left mouse button.
5. Help file information will appear in a window. This information is usually only summary procedural information on the particular topic. For more detailed information, refer to the actual Help file, and look for tutorial information on the topic of interest.
6. When done, click on the - button located at the top left-hand side of the Help window.

Note:

Under some circumstances, Method #2 will not activate the context sensitive Help file. When this occurs, use method #1.

The Help file search engine is broken down into two categories they are:

Search for Help On

This allows you to search for a topic or view the topics listing in alphabetical order.

Important Note:

Features which have not been implemented in your particular version of DC8/DC FORENSICS but which may be provided in different releases of the program will be described in italics.

About DC-Art

The following information is obtainable under the “About DC-Art” feature:

- Program Name and Version #
- Copyright Information including Copyright Dates
- Upgrade Contact Information
- Program Registration Information
- System RAM Size
- Hard Drive Disk Space Available for use on the temp drive
- Program Registration Name (User Name)
- Serial Number
- Registration Code

Section 3 - How To

Tutorials

Do you need help with a specific problem? This could be the place for you. Here, we provide you with procedures that may require more than one tool or one step to accomplish.

Demo .wav Files

You will find a number of demo .wav files in the following directories:
The path to navigate to these files is as follows:

For DC8 version 8.1

Using the Windows XP operating system:

<user>\MyDocuments\ DC8\Wavefiles

Using the Vista or Win7 operating systems:

<user>\Documents\DC8\Wavefiles

For DC Forensics version 8.1

Using the Windows XP Operating system:

<user>\MyDocuments\ DCForensics8\Wavefiles

Using the Vista or Win7 Operating systems:

<user>\Documents\DCForensics8\Wavefiles

Note: For the Win 7 operating system, the “Documents” folder appears in the “Libraries” folder.

Shortcuts to the demo wave files can be found in the “Diamond Cut Audio” program group. From the “Start” menu, choose “All Programs”, then “Diamond Cut Audio” then “Demo Wave Files” and select the desired demo file shortcut.

These files are intended to help you understand through experimentation the use of some of the features found in your software. Here is a listing and description of the demo .wav files and some of their possible tutorial uses:

demo1.wav This file is a raw transfer of a Edison Lateral Cut Disc (thin-cut) test pressing recorded circa the late 1920s. It is just a snippet of a song called “My Sin” and can be found in its entire form and completely cleaned up on the Diamond Cut CD entitled “The California Ramblers - - - Edison Laterals 2”. This demo .wav file contains a large number of clicks and ticks (and some hiss too) and thus is primarily designed to be used to demonstrate the operation of the various impulse filters in your software. In particular, you will find a preset titled “demo1.wave de-clicker” within the EZ Impulse filter. This preset will de-click that file, but you can also use this demo file to experiment with the various parameters within that filter. It can also be used in conjunction with the EZ Clean and the Expert Impulse filters for experimentation purposes.

RadioDemo.wav This file is a recording made on the East Coast of the USA of a German Broadcast as received on a short wave radio receiver at around noon time, EST. It contains a lot of hiss and the audio level varies a bit throughout. This .wav file is useful to show the multiple filtering/effect capability of the Diamond Cut Multifilter. To hear what it can do, just bring up the Multifilter and then bring up the preset called “Short Wave Radio Anti-fade and Cleanup”. You will see a lineup consisting of 5 filters and effects connected in cascade. Use the Preview button in the Multifilter to hear the result and use the bypass button to hear the unfiltered sound. Experimenting with the various filters and parameter settings contained within this preset in conjunction with this .wav file should prove instructive.

BigClickCracked78Demo.wav This .wav file is taken from an electrically recorded 78 RPM record which has a crack running from the outside rim all the way through to the center hole. It is a snippet of “The Blue Danube” and was recorded in the early 1950s. This demo file is intended to instruct you in the operation of the “Big Click” filter. Note that ratio settings of around 1.4 to 1.5 are effective and that the “thump” produced once per revolution of the record can be squelched by turning on the “De-Thump” feature.

HissBurstDemo.wav This .wav file has a hiss burst roughly 8 seconds into it. This hiss burst is used to demonstrate the various functions associated with the Direct Spectral Editing feature as well as the Bi-Modal Interpolation feature. You can highlight the hiss burst and experiment with the various features found in those filters.

dssdemo.wav This .wav file is a simulation of a loud bar room situation in which a male person is discussing a situation on the phone with loud rock music playing on the jukebox in the background. One side of this phone conversation is picked up by another nearby party in the bar via a “wire” that he or she is wearing. The recording is binaural in nature with one microphone located close to the jukebox and the second microphone hidden on a person and located in proximity to the person who is on the telephone. It is intended to demonstrate the usefulness of the DSS (Dynamic Spectral Subtraction) Forensics filter in terms of rejecting loud background music (or other types of coherent noises) from a surveillance recording. To hear this demo in action, bring up this file and then the DSS filter found in the Forensics menu. Then, bring up the DSS preset titled “dssdemo”. To compare the results, switch the filter between normal Preview operation and Bypass mode. You may want to increase the “Output” control on the right side of the DSS to hear the conversation more clearly.

Note: This demo .wav file is only available in the DC Forensics version of the software.

ForensicDemo.wav This file is a snippet of an air-check recording of the famous inaugural speech made by F.D.R. in 1933 in which he stated that “we have nothing to fear but fear itself.” This recording was captured in real – time over the radio on a Victor Pre-Grooved Record recorder at that time. It was recorded from a radio

transmission out of the New York metropolitan area and was received in Newark NJ via a combo Victor AM radio / disc record recorder machine. This file includes a combination of random and coherent noise signals (hiss and heterodyning) which are damaging to its intelligibility. Much of the noise that you hear is native to the original air-check recording. Other noises have been added in to demonstrate the de-noising capability of 8 filters/effects cascaded in the Multifilter. To evaluate this functionality after bringing up this file, go to the Multifilter and click on the preset titled “Forensic Demo Clean-up Filter”, and then Preview it. You will note that the voice is barely intelligible until you enable the preset. You can easily switch back and forth between the raw file and the enhanced version by using the “Bypass” checkbox in the Multifilter.

Note: This demo .wav file is only available in the DC Forensics version of the software.

Voice ID / Voiceprint Demo Files: Six sample .wav files are provided with the software to help you learn more about the voiceprint patterns of a female’s, male’s and child’s voice as recorded through high and low quality signal paths. These files consist of three persons saying the same “cue word” sentence. The low quality versions were recorded simultaneously with the high quality versions for ease of comparison using the Diamond Cut Spectrogram and Voice ID feature. Here are the files that are included:

Female Voice ID Test Sentence - High Quality.wav

Male Voice ID Test Sentence - High Quality.wav

Female Child (12) Voice ID - High Quality.wav

Female Voice ID Test Sentence - Low Quality.wav

Male Voice ID Test Sentence - Low Quality.wav

Female Child (12) - Low Quality.wav

The “Low Quality” files were recorded using a very low cost dynamic microphone manufactured circa 1970 and a commercial grade mixing board manufactured in the mid 1970s. The “High Quality” files were recorded with a studio grade large diaphragm condenser microphone in conjunction with a “dedicated” studio grade microphone pre-amplifier. All 4 of these files were digitized via a high-end soundcard. All of the files are sampled at 44.1 kHz with 16 bit depths.

Note 1: These demo .wav files are only available in the DC Forensics Audio Laboratory version of the software.

Note 2: The persons whose voices you hear on these demos were born, raised and live in Northern NJ, USA, in case you are trying to identify their accents.

Note 3: The Female Child's voice is that of a 12 year old.

Analog tape recording Transfer Tips

Analog tape recording was developed in Germany sometime in the early 1940's. This technology was brought to the United States sometime after the end of WW-II. Ultimately, tape recording replaced other methods of audio recording, such as direct to disc mastering, acetate and wire recording. Early recorders held the recording media (ferrous oxide) on paper tape backing. The grainy texture of paper produced discontinuities with the contact with the recording and playback heads, producing numerous analog artifacts. Later, more sophisticated backings were employed to get around that problem, but new problems were introduced that only showed up in subsequent years. Some of the problems that were introduced are of concern to the audio restoration engineer or hobbyist, since the effects of age on the tape backing may have degraded the material. Care in handling and pre-processing the tape should be considered when transferring old tape recordings to your hard drive for restoration. The following pre-processing and handling precautions should be taken depending on the tape medium.

- **Paper Backed Tape:** This type of tape was recorded monophonically. The signal is located in the center of the tape. Therefore, a full track playback head is optimal for the best reproduction. If this is not available, use the two inner tracks of a 4-track playback machine summed together. If this is not available, re-adjust the height (if you have the mechanical inclination to do so) of the playback-head until the tape players output signal is maximized. Also, adjust the azimuth for the best reproduction of the upper registers of the audio spectrum.

- **Acetate Backed Tape:** Acetate tape was used through the 1950's and halfway through the 1960's. If it is uncertain whether or not you are dealing with acetate-backed tape, hold the reel up to a light bulb looking through the layers of tape. If the view appears to be translucent, the tape is very likely to be acetate. This tape backing deteriorates over time, often giving off the odor of acetic acid (vinegar). This deterioration can cause playback problems, because the tape changes dimension, and often bows, not allowing it to pass over the playback heads of the tape deck evenly. This produces dropouts if not properly dealt with. The only way to deal with this problem is to increase the tape take up reel tension. Some sophisticated machines have controls or internal adjustments to facilitate this. Simpler machines will require more imagination on your own part. Another alternative is to install pressure pads on your machine's playback head. In either case, realize that the increased tension required to force the tape to pass evenly and flatly across the playback head will increase head-wear. But that may be a worthwhile price to pay if you are transferring a priceless tape for restoration and digital archiving.
- **Mylar and Polyester backed Tape:** Mylar and polyester backing provides a superior backing for magnetic audiotape. However, there were some manufacturing problems that were widespread in the US tape manufacturing industry using this technology in the mid to late 1970's over the 9-year period that followed. The problem involved a chemical breakdown of the binder, which affixes the oxide layer. This chemical breakdown process produces what has become known in the industry as "sticky shed syndrome". It leaves a residue on the tape path components of the tape player, and distorts the location of the oxide layer as it passes the mechanism. This produces a permanent degradation of the frequency response of the tape recording. Therefore, a pre-processing step is necessary if you believe that you have a tape from this era having a Mylar or polyester backing. The standard process is to bake the tape in a pre-heated industrial grade electric oven (between 120 to 125 degrees F) for around 4 to 7 hours (4 hours for ¼ inch and 7 hours for 2 inch tape). (The "H" fields

produced by an electric oven are not high enough in level to modify the position of the tapes magnetic domains.) An industrial convection oven is ideal for this purpose, but a conventional electric oven will work also. If you are very careful, you can use a home electric oven, provided that you do not trust its thermostat to establish the baking temperature. Do not use microwave ovens, and avoid gas ovens. Make sure that the thermostat is calibrated on your oven before proceeding. To do this, use a high performance reference thermometer located inside the oven in order to establish the temperature setting calibration. You must be sure that the temperature of the oven is stable before placing the tape therein. Allow the oven a good hour to stabilize in temperature before inserting the tapes. If multiple tapes are placed in the oven, they should be spaced away from each other by an inch or so. After the tapes have been baked, remove them from the oven and let them cool for about a day (24 hours). The tape(s) should be transferred to an archival media shortly after this process has been completed, since the tape(s) will begin to degrade quickly after performing this procedure.*

***Note:** Some plastic tape reels may deform using this process depending on their chemical make-up. To assure that this does not occur, it would be best to transfer the tape to a metal reel before proceeding with this process.

- **Cassette Tape:** Cassette tape transfers present special problems. Since the head gaps for cassette machines are very small, and the speed is only 1 - 7/8 inches per second, a slight miss-alignment of the playback head will produce a pronounced distortion of the stereo image and the fidelity of the upper registers of the musical scale. It may be necessary to use the "Time Offset" feature to correct tape azimuth miss-alignment problems. Refer to the File Conversions section of this manual for details.
- **Analog Tape Hiss:** Analog tape hiss can be removed using the Continuous noise filter. A "fingerprint" sample of the

noise will have to be taken before the noise can be removed. Refer to the Filters section of the manual for details.

Tape Dropout Repair (reel-to-reel)

One problem often plaguing old reel-to-reel magnetic tapes is tape warpage, which is one cause of tape dropout. Tape dropout is observed as a periodic loss of signal, especially in the high frequency portion of the audio spectrum. You can easily note if warpage is the source of the problem by observing the tape as it glides over the playback deck tape guides. If the tape moves evenly over the guides, then the source of the dropout may be lost magnetic coating. However, if the tape moves unevenly (wobbles) across the tape guides, the probable cause of the audio dropouts is tape warpage. The reason for the dropout is that the magnetic tape surface does not maintain good physical contact across the playback head thereby lengthening the magnetic coupling pathway (and increasing the magnetic reluctance) of the system. There are two solutions for curing this problem at the transfer stage of your project. If you have an old cheap tape recorder that used pressure pads, it will do a better job of transferring such a tape compared to a pro-machine which does not employ pressure pads. But, if all you have is a professional / high-grade machine, you can create your own homegrown pressure pad for the transfer. Follow these steps:

1. Remove the Tape Head Cover / Shield
2. Thread the tape through the tape path as you normally would do
3. Identify the play head (it will be the last head (right-most) in the tape pathway)
4. Obtain a small cotton ball and create a piece about $\frac{3}{4}$ inch square.
5. Cut a piece of Duct Tape to a dimension of about 2 x $\frac{1}{2}$ inches.
6. Affix the cotton ball to the center of a piece of duct tape
7. Wrap the tape with the cotton over the top of the head and down to the chassis of the machine, locating the cotton ball directly over the tape and the tape head.
8. You may have to experiment a little to obtain the proper tension for a clean playback, but this will dramatically reduce the dropouts of your old tape.

9. Replace the Tape Head Cover / Shield to minimize hum pickup during the transfer.
10. You will have to remove the tape and cotton ball and repeat this process after each tape side that is transferred in this manner.

Enhancing Reel-to-Reel Tapes Recorded at Slow Speeds

Sometimes, old reel-to-reel tapes were recorded at low speeds such as $\frac{3}{4}$ ips. These tapes lost much of the high end because the magnetic recording head used had relatively large gaps. These tapes can be brought up to a much higher level of performance by applying the following procedure in the prescribed order:

1. Check to be sure that the tape oxide side of the tape is facing towards the recording heads of your tape player. Sometimes a reel-to-reel tape has been accidentally re-wound with a 180-degree twist, causing the backing side to face the playback heads. The tape will play in this situation, in some instances, but with very poor frequency response and output level (the recorded information may be backwards).
2. Transfer the tape to your Hard Drive
3. Run the Virtual Valve Amplifier using the “Red Hot Jazz” preset.
4. Make the destination the source.
5. Run the Virtual Valve Amplifier again, this time using the “Bright & Brashy Brass” preset.
6. Make the destination the source.
7. The noise floor of the tape will now be substantially elevated, but so will the high end of the audio spectrum, so let not your heart be troubled. The next phase of the process will eliminate the noise leaving behind a clean “top end.”
8. Highlight a few seconds of the noise signature found at the beginning of the Destination .wav file.
9. Run the Continuous Noise filter to reduce the residual hiss. Go lightly with this filter to avoid digital artifacts.
10. Run the Dynamic Noise filter using the “ $\frac{3}{4}$ tape hiss attenuator” preset to further reduce hiss.

Archival Recording Philosophy & Methodology

The archival recording process should be approached differently than audio restoration recording. The goals are quite different between these two concepts. When producing an archival recording, one should strive to capture all of the information possible on the source recording and then store its media under controlled environmental conditions. When performing audio restoration, one is involved in the modification of that source material by eliminating noises and modifying the spectral distribution of acoustical energy so as to be more pleasing to the ear. The philosophy behind archival recording is that the resulting product should contain as much information as possible for future use, and this includes all of the noises, clicks, pops, thuds, hum, buzz and any other extraneous noises. These noises can always be removed at a later point in time without disturbing the master archive.

Applying any type of filter prior to producing the archival recording is a very bad practice. Transfer the recording either “flat” or via its known inverse equalization curve. If the inverse equalization curve is not known, the best fallback position is to transfer it flat. Also, if you are archiving a collection of monophonic recordings from a disk or cylinder source, it should be transferred by using a stereophonic magnetic phono cartridge. This way, both the left and right groove walls are recorded and maintained as individual entities, and since one may be in better shape than the other in certain areas of the transfer, you want to capture both. You should use the best equipment that you can afford for this task. Keep in mind that today’s technology will always be poorer than the technology of the future. And in some cases, the only material that someone in the future will have to work with will be derived from what you have done today. Even if this statement is not exactly true, when you take an analog recording and digitally archive it, you are stopping the chemical degradation process dead in its track at that particular point in time. Perfect clones of your digital master can be reproduced every ten years or so in order to insure that no further degradation occurs.

If you are working with old full track tape recordings, and do not have a high quality full track recorder available, use a half track machine and record both channels independently. You may find that one track contains some dropouts while the other does not. And even if both

tracks have drop-outs, sometimes you will find that the drop-outs do not occur simultaneously, which will allow someone in the future to fix the problem using a tool like DC8/DC FORENSICS.

Note: The digital archival recording standard established by the US Library of Congress is, at present, 96 kHz Sampling Rate in conjunction with 24 Bit Resolution.

CDR Prep from a Commercial Cassette Tape

1. Record side one of your cassette tape, creating one continuous .wav file at a 16 or 24 bit 44.1 kHz sample rate.
2. Record side two of your cassette tape, creating a second continuous .wav file at a 16 or 24 bit 44.1 kHz sample rate.
3. Using the Mute and/or Cut function found under the Edit Menu, eliminate excess lead in from the two .wav files. Do the same thing at the end of the .wav files to eliminate excess "dead-time."
4. Run the Continuous Noise filter to remove wideband noise such as hiss and rumble.
5. Enhance the recording by running any of the audio enhancement tools such as either of the equalizers, dynamics processor, reverb, or Virtual Valve amplifier.
6. Click on the CD Prep Menu.
7. Click on Find and Mark Silent Passages.
8. Check to be sure that the markers have located themselves correctly between "cuts." Adjust them if necessary.
9. Click on "Quantize for CD Audio."
10. Click on "Chop File into Pieces."

CDR Prep from a Vinyl Record

1. Record side one of your record, creating one continuous .wav file at a 16 or 24 bit 44.1 kHz sample rate.
2. Record side two of your record, creating a second continuous .wav file at a 16 or 24 bit 44.1 kHz sample rate.
3. Using the Mute and/or Cut function found under the Edit menu, eliminate excess lead in time and the "needle-drop" from the two .wav files. Do the same thing at the end of the files in order to eliminate the "needle-lift" and excess "dead-time."

4. Run the Impulse noise filter using vinyl mode on each file in order to remove ticks and clicks.
5. Run the Continuous noise filter to remove wideband noise such as tape hiss.
6. Enhance the recording by running any of the audio enhancement tools such as either of the equalizers, dynamics processor, reverb, or Virtual Valve amplifier.
7. Click on the CD Prep Menu.
8. Click on Find and Mark Silent Passages.
9. Check to be sure that the markers have located themselves correctly between "cuts." Adjust them if necessary.
10. Click on "Quantize for CD Audio."
11. Click on "Chop File into Pieces."

Characterize Frequency Response/ Equalize Audio Response (Forensics only)

The following describes how you can use your Diamond Cut software to characterize the frequency response of an audio system including the acoustical environment in which it operates. This method involves the use of random (white) noise in conjunction with the Spectrum Analyzer used in "Live" mode. The general theory is to stimulate the sound system and its acoustical environment with a known distribution of noise (constant energy per unit Hertz, in this case). Then, a signal representing a portion of the acoustical output of the system will be fed back into the software program via a microphone and a full duplex soundcard in order to measure the resultant response. The Spectrum Analyzer will display the real time result of this measurement. If the sound system has an equalizer in the signal chain, it can be used to "flatten out" the measured response while viewing the Spectrum Analyzer. The following equipment will be required to perform this measurement and/or system equalization:

Equipment:

1. High quality full-duplex sound card.
2. Flat responding Electret microphone (\$20.00 at most electronics dealers) and stand.
3. LIVE Software.

4. Pentium Computer (minimum 200 MHz).
5. Sound system (preferably with a graphic equalizer in the signal chain) to be evaluated.
6. Optional: In some cases you will need a small mixer with a microphone input if your sound card is not designed to handle microphone level inputs (in other words, the sound card only has a line level input).
7. Optional: 1/3rd octave 30 band or 5 band parametric equalizer inserted in the signal path of your sound system. (Please note that an octave-weighted 10-band equalizer is not particularly well suited for this application.) One of these devices is needed only if your goal is to equalize your sound system. If you are merely interested in characterizing your system, an equalizer is not required.

Setup:

1. Set the microphone on a stand a few feet in front of the listening spot most often used centered between the two speakers.
2. Connect the output of the microphone to the mic input of your sound card. If you do not have a microphone input, you will need a mixer having a microphone input to take the signal up to line level in order to be compatible with the line input of your sound card. In that case, the microphone would be connected to one of the microphone inputs of the mixer, and the line output of the mixer should then be connected to the line level input of the sound card.
3. Connect the output of the sound card to an auxiliary set of line level inputs on your sound system.
4. Defeat all tone controls on the sound system. If there is an equalizer in your system, make sure that it set for "flat."

Procedure:

1. Create a 10-minute stereo .wav file containing white noise. This is accomplished by using the "Make Waves" feature found under the "Edit" menu. Set the parameters as follows:
 - a) Wave shape: White Noise
 - b) Length: 600 Seconds
 - c) Amplitude: -10 dB

- d) File Type: Stereo
 - e) Sample Rate: 44.1 kHz
 - 2. Run the generator by clicking on "OK". After the computer performs this process, you should see a signal in the Source display. This process may take a minute or so.
 - 3. Make sure that the "Output Device I/O" is set to the high performance sound card in your system.
- *See the optional System Stimulus method outlined below.
- 4. Open up another copy of DC FORENSICS.
 - 5. Set the "Output Device I/O" of the second version of the program to a sound card other than the one that you are using for the test. This is found at the bottom of the "Edit" menu. Perhaps, choose the sound card that came with the computer for this setting since it does not have to be high performance.
 - 6. Set the "Input Device I/O" of the second version of the program to the high performance sound card in your system.
 - 7. Click on the "Live" button on the second version of the program.
 - 8. Click on the View Menu and bring up the Spectrum Analyzer.
 - 9. Set the spectrum analyzer as follows:
 - a) Display Mode: Slow
 - b) Range: 50 dB
 - c) Frequency Bands: 4096
 - d) Frequency Resolution: 2.69 Hz
 - 10. Activate the system by clicking on "Live Preview" in the Multi-Filter box.
 - 11. Using the "Offset Control" on the Spectrum Analyzer, center the signal on the screen.
 - 12. Play and loop the .wav file using the first version of the program that was used to create the .wav file. Adjust the volume on your sound system until the "hissing" sound is at a moderate listening level.
 - 13. Re-adjust the "Offset Control" on the Spectrum Analyzer for the best view of the signal.

The Spectrum Analyzer should now be displaying the frequency response of your system. The horizontal (x) axis indicates frequency and the vertical (y) axis indicates relative amplitude in dB. If the displayed signal is not flat and you desire it to be, you can engage your systems' graphic equalizer and adjust it until the spectrum display is as "flat" as possible.

***Optional System stimulus method:**

Rather than using two versions of the software program operating at the same time on your computer, you can create a CD ROM of the .wav file containing the white noise signal. Play this CD on your system CD player while using the spectrum analyzer operating in "Live" mode. This method has the added potential benefit of including your sound systems' CD player in the measurement results.

Convert White Noise into Pink Noise

1. Under the Edit menu, click on "Make Waves".
2. Select "Random" noise, with the amplitude set to -10 dB, and the length set to 5 or 10 seconds.
3. Click "OK", and a random white noise file will be created in the Source window.
4. Click on the "Filter" Menu and then the Paragraphic Equalizer.
5. In the preset menu at the bottom of the Equalizer, find and click on the "White to Pink Noise Converter" and then run that filter.
6. Pink noise will now be found in the Destination window.

De-clicking a Vinyl LP Record

This procedure makes use of the Expert Impulse Noise Filter. You may also want to try our new powerful EZ Impulse Noise Filter for quick and pain-free editing.

1. Download the Vinyl LP .wav file into DC8/DC FORENSICS using the "Open" command.
2. Highlight the portion of your .wav file on which you desire to apply the Impulse Noise Filter. (You may also choose to highlight the entire file.)
3. Click on the Filter Menu and then "Expert Impulse Noise"*.
4. Click on the "Vinyl LP" checkbox.
5. Set the Threshold Control to its minimum setting of 1.
6. Set the "Tracking" Control somewhere in the 25 to 30 range to start.
7. Set the "Size" control to a value somewhere in the range of 10 to 15 samples.

8. Run the filter or use Preview mode, and determine if the .wav file is being adequately de-clicked. If it is not, lower the tracking control. If it is de-clicking, but producing distortion on the sibilant sounds, then raise the control. Continue this process until a good balance is established of minimum sibilant distortion and minimum click feed-through. If it is de-clicking, but leaving a bit of an artifact behind, increase the value of the "Size" control.
9. When you determine the best setting of the control for your particular .wav file, click on Run filter. When the filter has completed its operation, the results will appear in the "Destination" workspace.

Important Note: Vinyl LP mode works best on .wav files that have been sampled at 44.1 kHz or higher. The setting examples given below are based on .wav files that have been recorded at a 44.1 kHz sampling rate only.

Important Note 2: Unlike all of the other DC8/DC FORENSICS controls, Vinyl LP mode cannot be turned on or off "live" in Preview Mode.

***Important Note 3:** Alternatively, you can use the EZ Impulse Filter which is very effective on Vinyl LPs, 45s, or 78s.

Manually de-clipping an over modulated Wave file*

Sometimes in the process of digitizing an analog signal into a .wav file, clipping can occur if proper attention had not been paid to the signal level during the recording process. Clipping is the process wherein the analog signal's amplitude is occasionally requiring more than the full-scale capability of the A/D converter in the recording system. This produces "flat-topping" of signals at their peaks that can yield a harsh sounding distortion upon playback if it happens too frequently. ***An automatic De-Clipping filter is now included with this product, but some prefer to perform the de-clipping function manually.*** The following is a procedure for manually de-clipping the sections of an over-modulated digital signal:

1. Open the clipped .wav file in the Source workspace.

2. Using the Gain Change algorithm (Edit Menu) reduce the amplitude of the signal by changing the gain of the entire .wav file by -6 dB.
3. The resulting areas of the signal that have been clipped will be obvious after you perform this operation by simple inspection of the Source display. The signals which had been clipped will appear as if they have had a "crew-cut"
4. Zoom-In on the areas that appear to have been clipped until you see the actual "flat-topping" of the problem section of the .wav file.
5. Highlight the area of the waveform that is "flat-topped." This process must be performed one clipped event at a time. You should not highlight a section of the waveform that has multiple "clipping" events for this process to work. Be very careful to highlight only the area that is clipped (flat-topped), making sure not to highlight too much of the signal on each side of the clipped section of the waveform.
6. Depress the "I" key (to interpolate the peak of the signal).
7. If you are de-clipping a stereo .wav file and prefer to apply the correction to only one channel, use "Ctrl-I" to interpolate the right channel only or the "Shift-I" to interpolate the left channel only.
8. You will see that the replaced signal has a rounded top rather than a flat top. The rounding of the flat top substantially decreases the distortion produced due to the clipping at that point in the signal.
9. Continue looking through the .wav file for all signals that are "clipped," repeating the procedure outlined above.

*Note: Alternatively, consider using the automatic de-clipper which often will do an effective de-clipping job on most material.

Decode Touch Tone Signals into Alpha-Numeric Values

1. Highlight the section in the recorded conversation containing the DTMF touch tone signals.
2. Play looped this entire section of the .wav file with the VU meter turned on (View Menu).
3. Using the Adjust Gain feature (Edit Menu), set the signals to indicate approximately -8 dB on the VU meter at the right of the screen while playing.
4. Bring up the Paragraphic Equalizer (Filter Menu).
5. Find the preset labeled DTMF Comb Filter (Normal)*.

6. Bring up the Spectrum Analyzer (View Menu).
7. Set the Spectrum Analyzer up in the following manner:
 - A. Slow Mode
 - B. Show Peak Enabled
 - C. Resolution = 0.67 Hz
 - D. Frequency Bands = 4096
 - E. Range = 100 dB
8. Highlight and then Preview the first (or following) tone burst using the preview button in the Paragraphic Equalizer‡.
9. Note that there are two major spikes on the Spectrum Analyzer. The peak detector will have automatically detected one. Click on the other spike so that a square box marker is following it.
10. Note that these two frequencies are displayed in the right and left corner of the Spectrum Analyzer display. Write down their values.
11. Continue this process (back to step 8) with the next tone burst, writing down each frequency pair.
12. When done, go to the DTMF chart in the glossary/appendix portion of this manual. From this chart, you will be able to decode the frequency pairs into a telephone number.

***Note 1:** Alternatively, you can select DTMF Comb Filter (Narrow) in very noisy situations. Also, you can choose DTMF Comb Filter (Ultra-Wide) in situations where the tone frequencies are off tolerance.

‡Note 2: The Forensics Brick Wall filter also has a special DTMF Band-pass Filter which can be used instead of the Paragraphic Equalizer when you encounter situations where there is an extreme amount of out-of-band noise which needs to be rejected.

Forensic Tape Authentication

Audiotape authentication is necessary if it is to be used in a legal proceeding. The process of making this determination is called tape authentication, meaning to determine that the tape specimen is the original.

Two separate "hum" signals existent on the media usually suggests the presence of a "dub." The reason for this is related to the imperfect nature of analog tape recorders that are commonly used in the forensics

world due to their small size. These recorders pick up small stray electromagnetic waves from the power distribution grid in proximity to the machine. An "original" recording will have just one noise signature at either 50 Hz or 60 Hz or harmonics thereof depending on the line frequency of the power grid in which the recording was made. Since analog recorders produce a certain amount of tape slippage, this frequency will be slightly offset from the actual line frequency. If a "dub" is then made of the master, a new noise frequency will appear on the specimen being analyzed with the spectrum analyzer (or the high definition Spectrogram). For example, if you see two spectral lines at, let's say, 60.1 and 60.4 Hz, this usually indicates that the test specimen is a dub. On the other hand, if you only see one spectral line, the tape is usually an original. To perform tape authentication utilize the following steps:

1. Make a Hard Drive copy of the tape on DAT &/or on your Hard Drive. CDR backup is also recommended.
2. Bring up the .wav file of the tape.
3. Bring up the Spectrum Analyzer (View Menu).
4. Set the FFT Size to 4,096 or greater.
5. Click on 0.67 Hz or smaller, and you will see that the system will re-scale the horizontal (X) axis on the spectrum graph to indicate 700 Hz full-scale.
6. Highlight a relatively quiet section of the .wav file, but one that is within the context of the recorded conversation.
7. Use the Range Control setting at 20 dB and the Offset control to zoom in on the signals of interest at around 50 or 60 Hz.
8. If there is only one signal around that particular frequency, then the tape is usually an original.
9. If there are multiple frequencies around the line frequency, then the tape is usually a dub.

Important Note:

The power grid in the United States can have an instantaneous frequency variation of up to +/- 0.5 Hertz for up to a 10 second interval. When measured over a 24-hour interval, the frequency tolerance is much tighter.

Automatic Micro-cassette Tape Start-Stop Sequence Detector (Forensics Version Only)

Forensics examiners are often confronted with the problem of authenticating a recording. Often, these recordings are of the micro-cassette variety. At issue is the question of whether or not the tape has been edited or modified in some way. Often, tapes are edited on a different machine from their original recording device. If a different machine was used to edit the tape, and since each machine leaves its own distinctive stop / start pattern when viewed in the time domain (display), it can be determined if the start / stop breaks were all made on one machine or two or more.

But, often the examiner is confronted with hours of tape and needs to focus in on the start / stop sequences rather than sitting through the entire tape(s) looking for these interruptions.

The “Microcassette Tape Start-Stop Detector” Multifilter preset will provide a way to Mark the file where these breaks occur so that a close examination of the file can be made in the time domain display without listening to the entire file(s) to look for recorder erase / record head signatures on the recording.

Here are two methods that one can use to operate this Multifilter Preset.

Method 1:

This method is the easiest to use, but is not self-documenting. In Classic Edit mode, run the attached preset on the Source file with the system (View Menu) set for Sync mode. Each start / stop sequence will be identified by a very large rail to rail pulse which shows up in the destination display. You can then zoom-in on the various start / stop events in the Source Display for further study.

Method 2:

This method is a bit more complicated to use, but it is self documenting (meaning that the audio material and the markers are integrated into the same wavefile).

1. Use the File Converter Filter to make a Stereo File out of the Mono one that you have from the Microcassette recording.
2. Make the Destination the Source.
3. Clone the File.
4. Run the above mentioned Multifilter preset on either the Left or Right Channel only.

Results: You will see a large pulse appearing on one channel wherever there is a Microcassette start / stop sequence. The pulse should be clear and obvious and rail to rail in amplitude.

Attenuating GSM Cell Phone Noise from Forensics Recordings (DC Forensics Only)

There are two common cell phone standard systems in wide use. One of the two, GSM (Global System for Mobil Communications) can cause interference to electronic equipment proximal to a cell phone (by a process involving Amplitude De-Modulation from non-linear devices interacting with Miller capacitance(s) in your recording device or other electronic system(s)). The high crest factor pulsing sound produced by these cell phones can overwhelm a Forensics audio recording making it indiscernible in its native format. Your software has a dedicated Cell Phone Noise Filter found under the Forensic Menu designed to attenuate this noise type. Refer to that section of this documentation for details. If that filter is not effective for your purposes, consider using the following alternative.

There are some Multifilter presets that may be useful for cell phone noise attenuation. Here is a procedure to try on recordings plagued with this type of cell phone noise that were not able to be reduced by the Cell Phone Noise Filter:

1. If the recording is not sampled at 44.1 kHz, 16 bit, then convert it to the stated format using the Change Sample Rate/Resolution feature found under the Edit menu.
2. Normalize the Gain of the file to -4 dB using the Normalize Gain Scaling feature found under the CD Prep Menu.
3. Bring up the Multifilter.
4. Find the Multifilter Presets having "Cell Phone Noise Interference Attenuator - x" as a prefix. You have 6 to choose

from, with each one in the numerical sequence being progressively more aggressive.

5. Preview each one and choose the one that performs the most effective noise reduction of the GSM cell phone noise without damaging the target audio signal.
6. If needed, adjust filters for the optimal results on the target audio signal.
7. Run the best filter.
8. Perform the usual post-processing steps to clarify the speakers voices using whatever your favorite Forensics filters may be.

Note 1: These Multifilters require a great deal of processing power. As a result, your CPU may not be able to keep pace with them in real time in “Preview” mode. Thus, you may hear some stuttering when previewing. This stuttering will not be heard after the filter has been “Run”.

Note 2: CDMA (Code Division Multiple Access) Phones do not cause this type of interference to other electronic equipment.

Security Tips for Forensics Audio Laboratories

Here are a few security tips one should consider when setting up a Forensics Audio Laboratory:

1. The laboratory should be secured in a manner in which only authorized personnel are permitted entry into the Forensics audio laboratory proper.
2. When unauthorized persons need to gain access to the laboratory area, written protocols should be established and strictly followed to allow clients and visitors entry into the laboratory including sign-in/sign-out sheets. But, these persons should never be provided with unfettered access to the lab; these persons should be made to leave the lab area when there are no authorized persons therein.
3. Unauthorized persons should never be left alone in an Audio Forensics Lab. This is very important and that is why we repeat it.
4. A security system should be installed on the laboratory portion of

your Forensics facility. This should include intrusion detectors on windows, passive infrared detectors, and a keypad or biometric entry system coupled to an electrically activated locking system on the labs entrance.

5. An independent computer system (not associated with your Forensics Audio Workstation) should be used to monitor the security system. This system should be hard-wire monitored by a central security service.

6. Video cameras located in the lab area are recommended and should be enabled 24/7.

7. Your Audio Forensics Workstation computer(s) should NOT be connected to the internet in any way. Wireless connections to these workstations should be disabled. You need to be able to assure clients and the legal process that files on these computers were not tampered with and one way. One way to reduce that possibility is to eliminate outside communications connections with your Forensics Audio Workstations.

8. Another major security risk is portable USB drives. Extreme care needs to be taken when using an portable USB drive. These drives can carry virus files to your system and be used to extract critical information.

9. Your Forensics Audio workstation(s) should not be networked with computers outside of the audio forensics laboratory proper.

10. Keep only software programs that are absolutely necessary for your Forensics Audio work on your Audio Workstation. Billing and admin computers should be located elsewhere in the facility and not networked with the Forensics audio workstation(s).

11. Software registration should be performed manually. Use an admin computer outside the lab to obtain the necessary registration codes for your software and "sneaker-net" that information to your Forensics Audio Workstation(s).

12. Master (originals) audio materials should be stored in a fireproof

safe. Backup copies of these audio materials should be kept in a secure area off-site. Audio materials should always be kept under lock when not being transferred to your Forensics Audio Workstation Computers.

13. Client Audio materials should never be left unattended outside of the laboratory proper.

14. Master hard-copies of your audio restoration software should be kept in the laboratories fireproof safe and this safe should be locked when the hard copy materials are not needed.

15. Cell phones should not be allowed to be used within a Forensics Audio Laboratory. They should be turned off (not just silenced) or left outside the laboratory proper. This rule applies to authorized and non-authorized visitors to the lab.

Forensics Audio Handling & Chain of Custody

If you are dealing with Forensics Audio materials or evidence, certain protocols need to be followed. There are several reasons for this:

1. The need to protect the evidence from potential tampering by third parties.
2. The need to prevent accidental damage from occurring to the materials while in transition or while they are in your possession.
3. The need to survive legal scrutiny under examination in a legal venue pertaining to the proper handling of the materials.

In terms of maintaining a traceable “Chain of Custody”, the following procedures should be followed:

1. Shipment to or from a client should be traceable via Registered or Certified US Mail.
2. A packing slip or letter should be included with the shipment describing the material enclosed.

Certain precautions should be followed when shipping Forensics Audio

materials to assure its integrity throughout the process. The following recommendations should be followed:

1. Do not ship the audio materials with any other non related items.
2. 6 inches of packing material should be provided around the extreme dimensions of the media. This is to minimize the effects of shock and magnetic fields on tape based materials.
3. Use only non-shedding packing materials such as bubble wrap. Do not use shredded newspaper or anything similar since fibers can negatively affect tape mechanisms.
4. Use anti-static packing materials whenever possible.
5. Keep a written and photographic record of the packing process used and maintain that in a document folder for your clients project.

When you receive Forensics Audio materials from a client, it is advisable to follow this procedure:

1. Immediately inspect the outside of the package for damage. If there is any damage to the package, photograph it and document your findings in writing and place that documentation in the clients project folder. Immediately notify the carrier of the damage noted on the shipping material and inform your client of the damage noted.
2. Open the container and inventory the contents found therein. Compare the contents against the packing list or letter. If there are any discrepancies, note these in writing, place those notes in the clients project folder and inform your client of your findings.
3. Save all shipping labels & documents attached to the shipping materials/package.
4. Carefully inspect all packing material to be sure nothing of value is discarded.
5. Photograph the recording(s).

6. Date and mark the recording(s) for identification.
7. Conduct a detailed physical examination of the recordings. Points of interest should be photographed. Keep detailed written notes as you examine the materials and maintain those notes in the client's project folder.
8. If not already removed, remove the safety recording tab from Cassette tapes before playing. Do not discard the tab. Put it in an envelope, seal it and document its contents and keep this in the client's project file.

Use the following guidelines whenever handling a clients audio evidence:

1. Use Gloves whenever handling Forensics Audio materials. Only handle the material when absolutely necessary for examination, playback, or return to the client. Avoid any unnecessary handling of the material and be sure not to touch the actual media recorded surface with a bare hand or ferrous tool.
2. Create a direct digital copy of the material. Perform all of your analysis using the digital copy rather than working continuously with the original. Using the Diamond Cut recorder, transfer the recording with 24 bit resolution and a 96 KHz sampling rate. Keep a written record of the time that the recording was transferred and the file name assigned. Make a backup copy of the digital recording on CD or DVD media, and keep this copy in a secure location off-site.
3. Always be sure to keep magnetic media away from all sources of magnetic fields such as computer monitors, loudspeakers, permanent magnets, power tools, etc..

When you are not using the Audio Evidence, you should abide by the following rules:

1. Limit access to the material to those in your laboratory who have a need to know based on their involvement with the project.
2. Keep the material in a fireproof Safe which is kept locked.

Preferably, this safe should be located in the basement of your building near a 90 degree corner of a foundation wall.

3. Assure that the material is kept in a constant temperature and constant humidity environment, keeping it stored away from any source of magnetic fields.

Gain Riding Procedure

1. Highlight the portion of your .wav file during which you would like to "slew" (change) the gain up to or down to a new value. (If this is to be done on a relatively small portion of the .wav file, it may be useful to Zoom-In on the sector of interest.)
2. Press the Play button to audition your selection.
3. Click on either "Fade-In" or "Fade-Out." (Edit Menu)
4. Choose the type of slew curve you prefer, either Linear or Logarithmic.
5. Set the "Start Level" slider control to unity gain.
6. Set the "Stop Level" slider control to the desired new gain for the segment of interest. This can be up to + 6 dB (amplification) or up to - 96 dB (attenuation).
7. Click on OK. The gain "slewing" will be performed during the highlighted portion of the .wav file.
8. Highlight the portion of the .wav file that you desire to remain at the new gain setting. This will be the "dwell" sector for the gain riding procedure.
9. Again, click on either "Fade-In" or "Fade-Out."
10. Next, change the "Start Level" slider to the same value as the "Stop Level" slider. This will be your new gain setting (dwell).
11. Click on OK. The gain modification will be applied to the highlighted portion of your .wav file for its highlighted duration or dwell time.
12. Next, select the portion of your .wav file during which you desire the gain to slew back down to its original value (or another new value).
13. Again, click on either Fade In or Fade Out.

14. Change the "Stop Level" slider back down to unity gain (or whatever new gain you desire) leaving the "Start Gain" setting where it had been.
15. Click on OK. The gain will slew back to the unity gain value during the highlighted time interval.

De-Clicking with the Impulse Filter and "Sync Mode."

1. Place DC8/DC FORENSICS into "Sync" Mode (View Menu).
2. Automatically De-Click the entire .wav file in the standard manner utilizing the Expert or EZ Impulse Filters.
3. Listen to the result in the Destination Workspace. Listen for locations that contain pops or clicks which were too severe for the algorithms to conquer.
4. Zoom-In on the pop or click that you desire to remove.
5. Lower the setting of the Threshold Control (or the Tracking Control if you are working with Vinyl LPs), and run the filter. Observe in the Destination Workspace to see if the increased sensitivity was able to remove the impulse.
6. Keep repeating the above process until you are satisfied that the impulse pop or click has been removed and replaced with a reasonable looking and sounding replacement waveform.

Impulse Noise Generation

Sometimes it is desirable to be able to generate Random Impulsive Noise. One application involves room acoustical propagation delay testing. Another application is theatrical in nature; that is to say that a modern recording may need to have some "aged patina" added to give it the sound of a time gone by and contemporaneous with the time of a particular theatrical performance.

The following Presets can be found under the Multi-Filter for Random Impulse Creation:

- White to Impulse Noise Converter 1 (45 RPM Low Cost Record Player Simulator)
- White to Impulse Noise Converter 2 (33 RPM Low Cost Record Player Simulator)

- White to Impulse Noise Converter 3 (45 RPM Hi-Fi Quality Record Simulator)
- White to Impulse Noise Converter 4 (33 RPM Hi-Fi Quality Record Simulator)
- White to Impulse Noise Converter 5 (Radio Static Simulator / Acoustical Test Signal)
- White to Impulse Noise Converter 6 (78 RPM Record Simulator *)

The first set of 5 presets will allow you to generate Random Impulses using the DC Forensics version of the software. You will need to start with a "Makes Waves" file consisting of Monophonic, 44.1 kHz, 16 bit Random noise at the -10 dB level. Next, apply the resultant signal to any of the first five presets in your Multi-Filter and the result will be randomized impulses of varying characteristics. This signal can be used to simulate an olde recording by "Paste Adding" it to a modern recording. (Please refer to the Bandpass Filter preset listing for additional theatrical sound simulations.) If you do not have the DC Forensics version of the software, download the DC Forensics / Forensics demo from our website located at www.diamondcut.com and use it to create the file that you desire. If you need more than a one minute duration of random impulses, use the concatenate function in your DC EIGHT software to elongate it.

*The sixth preset (78 RPM Record Simulator) requires a different stimulus signal that can be created using the following procedure:

1. Create a Make Waves file of Random Noise, - 10 dB, 16 bits, 44.1 kHz, Mono, 30 seconds long (which represents 39.00 revolutions of the record).
2. Copy that file to the clipboard.
3. Create a Make Waves file of Sine waves, 1.3 Hz, -30 dB, 16 bits, 44.1 kHz, Mono, 30 seconds long.
4. Paste Mix this file to the one on your clipboard.
5. Apply the White to Impulse Noise Converter.
6. Preview and/or Run that filter preset on the file that you had just created.

The resultant file, (or concatenated groups of files) when added to a musical source, will give you an excellent simulation of 78 RPM record

crackle which can be useful if you are trying to create an artistic effect for theatrical purposes.

If you need to perform acoustical room propagation delay testing, use White to Impulse Noise Converter 5. If you only want one impulsive event every 10 seconds, use the Edit/Mute function to establish that relationship. If you need to modify the shape of the impulse(s), Zoom-In and use the Pencil Tool to make the desired changes.

Note: If you desire a stereophonic simulation of Impulsive Noise, simply "Make Waves" in stereo instead of mono. Then, use the file conversion filter to create stereo using a time offset value of 20 mSec before applying the stimulus signal to the White Impulse Noise Converter preset of choice.

Nudging the Highlighted portion of the workspace

1. To Nudge the right-hand side of a highlighted portion of a DC8/DC FORENSICS .wav file, merely use the left and right arrow keys on your keyboard.
2. To Nudge the left-hand side of a highlighted portion of a DC8/DC FORENSICS .wav file, depress the Shift key while operating the left and right arrow keys.
3. To change the resolution of the Nudge feature (the number of samples per nudge) go the preferences section of the Edit Menu, and enter the desired value of samples for each nudge.

Stanton 500 RIAA Compensation Curve Preset for the Multifilter

Under the Multifilter you will find a preset called "Stanton 500 RIAA Compensation Curve". This preset is to be used in conjunction with a Stanton 500 cartridge and a "Flat" phono preamplifier. This preset will not only convert a flat transfer using a Stanton 500 to the RIAA curve, but will also compensate for deficiencies in the flatness of response of the phono cartridge itself. This was accomplished through the use of a test record made by "Hi-Fi News" in conjunction with the Diamond Cut Spectrum Analyzer. The Stanton 500 was chosen for compensation because it is very popular, readily available, relatively low in cost, and can be outfitted with LP and 78 styli. It is important to emphasize that

this preset is not to be used with a transfer through an RIAA preamplifier, but only via a Flat preamplifier such as the CTP series.

Record Transfer to Hard Drive Technical Hints

The first order of concern when transferring records to your hard drive should be your system setup.

- Your turntable, since it most likely will have a magnetic cartridge, should be kept at least three feet away from your monitor, because the stray electromagnetic fields created by the monitor's deflection circuits can be a source of noise entering your pickup.
- Another important consideration is to be sure that your turntable's chassis is grounded to your pre-amplifier chassis through an independent grounding wire (not the phono cartridge shielded cables) in order to minimize hum pickup.
- Shielded cables should be used for all audio interconnections with the exception of the loudspeaker connections to your power amplifier.
- All power supply cords should be fed from the same wall outlet. This is done in order to minimize any possible ground loops that could occur if this technique is not followed. In other words, your computer including its monitor and printer, and your turntable, pre-amplifier, audio power amplifier and any other system accessories should all be plugged into the same multiple outlet strip. The outlet strip is then plugged into a source of power, such that all of your equipment is operating off of the same wall outlet. This is especially important when your audio equipment is of the three-wire variety (with a safety ground). This will be encountered more often with professional grade audio equipment as opposed to consumer grade equipment. If this technique is not followed, a ground loop may be created between your computer's ground connection and your audio system ground connection. When this occurs, a noise current will flow in your audio cable shields with a consequential noise Voltage drop appearing across the same. This can induce a hum and / or buzz into your recording. If it is not possible to operate all of your equipment on the same electrical outlet, then you should consider using an audio isolation transformer between your

audio pre-amplifier output and your computer sound card input to break this ground loop.

- When transferring records utilizing a magnetic phono cartridge, it is useful to "equalize" the frequency response of the pre-amp system before transferring the sound to your hard drive, unless the original source material was acoustically mastered. There have been many different equalization curves (RIAA, Decca, etc.) employed by the various record manufacturers. The primary purpose of these various equalization schemes was to allow for a fuller bass response without producing an over modulation of the record groove wall. The frequency below which the cutter system reverted to a "constant displacement" as opposed to a "constant velocity" is known as the turnover frequency. There are pre-amplifiers available which allow you to select an appropriate turnover (and roll-off) frequency depending on the brand of the record that you are transferring. The turnover frequency for most electrically recorded 78-rpm records generally falls in the 200 Hz to 500 Hz area. If this is not accounted for, your transfers will sound "thin" on the bottom end. In addition to turnover, LPs used some pre-emphasis of the frequencies above around 5 kHz. If a Roll-off is not utilized during playback, these records will sound harsh on the top end. The industry standard equalization that encompasses both a turnover and a roll-off characteristic is the RIAA curve. It was used almost exclusively on all LP recordings after 1955. Other equalization curves for LPs that were employed before 1955 include the NAB, AES, FFRR, and the Columbia contours.
- Although you may be transferring monophonic 78-rpm recordings to your hard drive, consider making the transfers in stereo mode. By capturing both groove walls of the recording, you can take advantage of some of the information captured in this way in order to reduce noise, and clean up the muddiness often found in the "bottom end" of old 78s.
- Make sure that the turntable is sitting on a surface having very little vibration coupling in from other mechanical sources. The best place to locate your turntable would be on its own table sitting on your concrete basement floor, or better yet, on the floor itself. When you make the transfer, turn off your

sub-woofers and keep the speaker volume very low in order to minimize acoustical feedback (and the resulting resonance) from appearing on the digital recording. Remember, after the transfer, you can play it as loud as you prefer without affecting the sound quality. One of the great benefits of digital audio CDs as a media compared to analog discs is the lack of an acoustical feedback pathway having a direct effect on the sound quality. Acoustical feedback is not a first order effect in the playback of digital recordings, but it is with mechanical analog records. However, we would not recommend placing your CD player directly on top of your loudspeakers, as this can cause errors or skips to occur at very loud listening levels.

- A very simple noise reduction technique involves conversion of the file to a mono file by adding the two files together and dividing the amplitude in one-half. This tends to cancel out rumble and low frequency noise.
- When transferring Hill and Dale (Vertically Cut) records such as Cylinder recordings and Edison Diamond Discs, no useful information can be extracted from a stereo representation, so save the disc space and transfer these monophonically. However, your pre-amplifier must be placed in subtractive mode in order to derive a decent signal for the transfer (left channel minus right channel). Since not all pre-amplifiers have this function, you may have to transfer to DC8/DC FORENSICS in stereo, and perform a stereo to mono L/R file conversion in software. Another method for obtaining the left minus right signal for the transfer of Hill and Dale recordings involves a slight modification to the tone-arm shell of your turntable. You can re-wire the stereophonic phono cartridge output terminal to be in series. The phasing must be arranged such that the two positive (hot) signals are wired together forming a node, and the actual output should be taken from the two negative (ground) terminals of the cartridge coils. The tone-arm shell should be connected to only one of the two cartridge negative terminals. Sometimes one of the two negative terminals is connected to a metal shield around the cartridge. This is the terminal that should be connected to the shielded conductor of one of the tone arm co-axial cables. It is important to note that when a stereo cartridge is connected in this manner, the output impedance will double and therefore

the manufacturers recommended value of load resistance should also be doubled.

- If you transfer acoustically mastered recordings through a magnetic cartridge that is driving an equalized pre-amplifier, you will have to apply the "Reverse RIAA" curve found under the Paragraphic Equalizer. If you omit this step, you will overly amplify the bottom end of the surface noise spectrum of the record. It will do very little to enhance the bass response of the transfer, and will cause the transfer to sound "muddy." When using a magnetic cartridge to transfer such records, the pre-amplifier equalization circuits must be disabled. If this is not easily accomplished, try using a Mic input for the magnetic cartridge, rather than the equalized magnetic cartridge input.
- The Mic Input on a preamp is usually fairly high in gain, and flat in frequency response. However, if the input impedance is higher than the recommended load resistance for your phono cartridge it will be necessary to terminate the input to the Mic pre-amplifier with an additional parallel connected resistor. Use the following equation to determine the value for this input shunt resistor (this equation assumes that the input impedance specification for your input is actually represented by a simple resistance in the audio frequency band, rather than a complex impedance, which is generally the case):

Note: If the input impedance of your microphone pre-amp is less than the recommended cartridge load impedance, this technique will not work.

$$1 / R_{\text{shunt}} = 1 / R_L - 1 / R_{\text{in}}$$

whereby - -

$1 / R_{\text{shunt}}$ = The reciprocal value of resistance you must add in parallel with the input of your system given in Ohms.

&

$1 / R_L$ = The reciprocal value of required load resistance specified by the manufacturer of your magnetic phono cartridge given in Ohms.

&

$1 / R_{\text{in}}$ = The reciprocal value of input impedance for the input which you will be using given in Ohms.

- Also, something worth considering (but not as important as load resistance matching) is providing the proper value of load capacitance for your phono cartridge. If capacitance loading is incorrect, ticks and pops on the record may produce a ringing waveform that could make it more difficult for the Impulse Noise Filter detector to discriminate between sound transients and noise impulses. When analyzing the value of load capacitance, consider the cable capacitance as part of the total. For example, if the cartridge manufacturer recommends a 250 pF load capacitance value, and the cables already have 150 pF (6 feet of cable at 25 pF per foot), then you only need to add 100 pF across the load resistor at the amplifier input.
- Avoid the use of special effects such as reverberation, graphical equalization, notch filtering and so forth at this point in the sound restoration process. The added complexity of these signals will make your job of sound restoration more difficult, because DC8/DC FORENSICS will have a tougher time of separating signals from transients and noise. These special effects, if desired, should be added after the signal restoration process has been completed. And lastly, do not cut off the top end of the signal bandwidth at this early stage in the restoration process. Some of the algorithms make use of the fast rise times of the transient signals in order to perform their function. The use of analog Low-pass filters at this stage of the process can create severe distortions of the ticks and pops on the source material. The "ringing" created by these filters can make it more difficult for the Impulse filter to perform its function. It is always easier to remove bandwidth from a signal source than it is to restore it, so this job is best left to the point in time where you are actually performing the sound restoration with the DC8/DC FORENSICS tool-set.

Removing a lead vocal from a Stereo Recording

The following procedure will very often be effective in removing a lead vocalist's performance from a stereophonic recording. This is useful when you desire to over-dub your own rendition of the performance onto the original song. If there is a lot of ambient sound (reverberation) associated with the original artist, it may not be removed along with the vocal itself. This is seldom a problem, since the ambient sound is not

very distinct, and will not drastically deter from your own over-dub performance. It is important to note that this technique will not work with a monophonic source. It is also important to note that this technique very much relies on the type of material you are using. It will produce better results on more acoustic type music and sometimes no success on music with heavy walls of instruments.

1. Open your desired stereo file.
2. Open the Channel Blender feature (Effects Menu).
3. Set Left and Right to 100% blend
4. Select Invert Phase on the left channel
5. Select Blend to Mono and choose "Below".
6. Set the Blend to Mono adjustment to 160 Hz.
7. Click on the Preview button.
8. You should hear the Source .wav file with an attenuated lead vocalist, with perhaps a little bit of reverberation (echo) in the background.
9. Adjust one of the two-slider controls downwards first. Observe if the lead vocalist gets louder or softer. If it gets louder, readjust the control upwards, looking for a point on the slider control wherein the lead vocal is nulled out (minimized).
10. Click on "Run Filter."

Note: This technique works using the same methods as hardware boxes costing many times the price of this product. However, it has the same limitations as many of these hardware solutions in that the vocals to be removed must have been mixed in the exact center of the stereo field by the recording engineer. Many recordings are made this way, but some are not.

Restoring an Old 78 RPM Recording: In Depth

The following are the steps in the process that we have been using for the restoration of early disc and cylinder recordings. You may find it useful to learn from our experience before undertaking an audio restoration of an old recording on your own. Please note that although most of these steps are based on 78s, the principles can usually be applied to all types of record restoration projects, including LPs.

1. Backup First

When dealing with a one-of-a-kind or a very rare recording, it is strongly advised that you make a transfer of the recording before beginning any cleaning process. The reason for this is so that you at least have something to work with in the event that you inadvertently damage the recording in the cleaning process.

2. Clean the surface of the recording:

Use a machine designed for this purpose if you have one available. The type that deposits a "bead" of distilled water and then removes it with a stylus, string, and a vacuum system are probably the best for this purpose. If a system such as this is not available, clean your record with a lint-free cloth and distilled water. Avoid the use of solvents or wetting agents which are non-aromatic* as they have the propensity to leave behind a residue. These residues can attract particulate matter over time and clog the bottom of the record groove. If you are cleaning either wax cylinders or Edison Diamond Discs, or other records containing wood or paper cores, do not use water because of the potential for damage by the solvent. Use only a lint free cloth on these items. Also, be very careful not to get fingerprints on wax cylinders. The oil in your fingerprint will provide the "seed" necessary to trigger fungus growth on the wax surface. This will destroy the cylinder groove wall in time. Blue Amberol cylinders can be cleaned with a cloth that has been moistened with distilled water. However, be careful not to allow any water to come into contact with the plaster core, because it may swell up cracking the record surface.

Important Note:

Do not use solvents such as alcohol or acetone on acetate (transcription) recordings! These solvents will destroy the recording. We have seen grown men cry after utilizing this method of cleaning on acetates (and in one case the transcription was a one-of-a-kind recording of a once famous German opera star).

***Note 1:** Aromatic solvents such as ethanol, methanol, or isopropyl alcohol diluted with distilled water are sometimes used to clean vinyl recordings. Never use benzene, gasoline, naphthalene or any other similar hydrocarbon based solvent to clean the surface of any type of recording!

Note 2: Acetate recordings are often covered with a white coating that appears as a powdery substance on their surface. This material (hexadecanoic acid) is not soluble in H₂O. It is suggested that records with this problem be played "wet" using distilled H₂O for the best transfer. Do not attempt to use solvents to remove the acid.

3. Play the record in a "dry run"

Play the record once on your turntable using one of your smaller tip stylus (approx. 2.3 mils). Obviously, you do not need to be able to listen to the recording during this process, and so you may want to consider using a separate turntable that you will use exclusively for this purpose that is not necessarily connected into your sound reproduction system. The purpose for this step is to kick up any accumulated dirt located at the bottom of the groove before re-cleaning the record surface.

4. Clean the Record Surface Once Again

Clean the surface once again using the procedure outlined in step #2.

5. Set your Pre-amplifier to the Proper Mode

Set your pre-amp to stereo mode for lateral cut 78 RPM records. This is not done because your 78 record was recorded in stereo, but the left and right groove walls will both be recorded independently utilizing this technique. Later, using some of the file conversion options available in DC8/DC FORENSICS, you will make some decisions regarding the best way to use this left and right groove wall information. If you are transferring vertical cuts (hill and dale) like Edison Diamond Discs or cylinder records, you will either have to place your pre-amplifier in Stereo (Left - Right) mode (which may be difficult since not all pre-amplifiers have this feature) or you will have to record in stereo, and use a file conversion feature to convert the signal to vertical using DC8/DC FORENSICS. If your pre-amplifier does not have the feature, the record will sound a bit noisy and weak as you listen to it. However, often when the file conversion is performed, the gain will increase and the signal-to-noise ratio will improve at that point in the restoration process. Make sure that your pre-amplifier is providing the full audio bandwidth to the digital recorder. The DC8/DC FORENSICS program will need as much bandwidth later to perform its magic. Do not concern yourself with the theory of

producing aliasing as some have suggested, since that problem has been solved by all reputable sound card or digital recorder manufacturers.

A flat pre-amplifier can be used such as the CTP series. The Diamond Cut Virtual Preamplifier can impart the correct equalization curve on the recording after the transfer has been made if that step is necessary. EQ is only required for electrically recorded records. Acoustically recorded records do not require any equalization.

If you are using a RIAA preamplifier to transfer 78s, you can use the Diamond Cut Virtual Pre Amp to reverse the RIAA curve and impart the correct Turnover curve to your transfer. Please refer to the section of this users guide pertaining to the Virtual Preamp (VPA) for details.

6. Verify that your Turntable Speed is Correct

Verify that the speed of your turntable is correct. Your Diamond Cut software includes a set of Strobe discs in the help file to help with this task. Just print the desired disc and cut it out to be placed on your turntables platter. A fluorescent light will be needed to illuminate the disc properly. Most Victor, Columbia, and other 78s were actually recorded at 78.26 RPM, however, Edison laterals were recorded at 78.8 RPM. Edison Diamond Discs were recorded at 80 RPM. Use a strobe disc and a fluorescent or neon lamp operating off of your AC line for this purpose, provided that you live in an area where the power line frequency is accurate and stable. If the speed is incorrect, use the turntable pitch control to correct the anomaly. If you do not have a pitch control on your turntable, often you can correct the speed later using the Diamond Cut Speed Change effect.

Note whether or not the record groove is rotating concentrically. If it is not, there will be a "Wow" effect on the recording. This problem can be corrected if your turntable has a removable spindle. With the spindle removed, adjust the position of the record with respect to the turntable platter until the stylus tracks the record concentrically. If your turntable does not have a removable spindle, AND the record is not a priceless artifact, you might consider using a ream to enlarge its centering hole. Then, with the stylus playing the record, bump the edge of the record lightly with your finger until your turn arm is operating concentrically. We must repeat - - - DO NOT use the ream method on an important artifact!

7. Verify that you are Utilizing the Correct Equalization Curve

Verify that you are utilizing the correct equalization curve for the particular type and brand of record that you are about to transfer. Turnover and sometimes Roll-off are critical breakpoint frequencies that must be matched in a complementary manner to the recording process in order to preserve the "flat" response of the original recording session. Turnover frequencies for electrical recordings are between 200 to 629 Hertz. Acoustical recordings should always be transferred "flat" and "electricals" should be transferred with equalization that is the correct inverse of the recording equalization that was used in the mastering process. Also, it is important to have a pre-amplifier that has the ability to adjust the turnover and Roll-off frequencies or to use the Diamond Cut Virtual Pre Amp to apply the correct EQ. More esoteric EQs can be found in the preset listing under the Paragraphic EQ filter. For more information on this topic, refer to the section entitled "Record Transfer to Hard Drive Technical Hints." Below is a list of common Turnover Frequencies for some of the more common brands of lateral cut 78-RPM records:

Type, Brand, or Process	Turnover Frequency
Acoustical Recordings	0 Hz
Columbia (1925 - 1937)	200 Hz
Victor (1925 - 1937)	200 Hz
Westrex	200 Hz
Decca (1935 - 1949)	250 Hz
EMI	250 Hz
English Columbia	250 Hz
HMV (1931)	250 Hz

EMI (1931)	250 Hz
London	250 Hz
Blumlein	250 Hz
Columbia (1938 – End)	300 Hz
BSI	350 Hz
Capitol	400 Hz
Mercury	400 Hz
Brunswick	500 Hz
Decca (1925 – 1929)	500 Hz
Edison Laterals (1929)	500 Hz
MGM	500 Hz
Parlophone	500 Hz
Victor (1938 – 1952)	500 Hz
629	

8. Choose the Best Stylus

Choose the best stylus for the record transfer. In most cases, truncated elliptical styli are the best for transferring olde 78s, since the stylus tip will not be in contact with any dirt and grit at the bottom of the record groove. The criteria for stylus selection should be based on two parameters. The first is signal-to-noise ratio (in more common terms it would be referred to as surface noise). The second is distortion. Keep in mind that it is always possible to improve the effective signal-to-noise ratio of a recording, but it is much more difficult to decrease the harmonic distortion content of a recording in any post-transfer process.

So find the stylus that produces the lowest distortion as the first criteria, and listen for the stylus that produces the best signal-to-noise ratio as secondary criteria. Styli are available in different diameters specified in mils, and in geometries such as spherical, conical, and truncated versions of both. Although there are charts which call out the best stylus to use for a particular record brand and era, you will probably find that the best one is always determined by trial and error, since the charts have no way to account for the wear pattern of the particular disc

which you desire to transfer. However, if you prefer a more deterministic approach, you can use the following styli list to get into the right ballpark:

1. **Edison 80-RPM Diamond Discs:** 3.7 mil spherical or non-truncated conical stylus.
2. **Wide Groove Acoustical 78 RPM Lateral Discs:** 3.8 mil truncated elliptical stylus.
3. **Edison White Wax, Brown Wax, Concert, and Gold Molded:** 7.4 mil Spherical stylus.
4. **Edison Blue Amberol Cylinders:** 3.7 to 4.2 mil non-truncated spherical stylus
5. **Edison Wax Amberol Cylinders:** 4.2 mil spherical stylus.
6. **Pre-1935 Lateral Cut Electrical 78s:** 3.3 mil truncated elliptical stylus.
7. **Transcription Recordings:** 2.3 mil truncated elliptical stylus.
8. **Late 1930s Lateral 78 RPM Discs:** 2.8 mil truncated elliptical stylus.
9. **Narrow Groove 78s such as Polydor:** 2.4 mil truncated elliptical stylus.
10. **Standard Groove 78 RPM Discs:** 3.0 mil truncated elliptical stylus.
11. **Modern LPs:** 0.7 mil elliptical stylus.
12. **Early LPs:** 1.5 mil truncated elliptical stylus.
13. **1931 to 1935 RCA Pre-Grooved Home Recordings:** 5.0 mil spherical stylus
14. **Pathé 78s:** 3.7 mil truncated conical stylus
15. **Aluminum Instantaneous Discs:** 6.0 mil conical
16. **Etched Label Pathé to 14 inches in diameter:** 8.0 mil conical
17. **Etched Label Pathé over 14 inches in diameter:** 16 mil conical

Important Note:

Actual Groove widths can be measured with a 200X to 300X microscope equipped with a calibrated reticle. The location of the wear pattern can also be observed, so that you can choose a stylus having a dimension that will either ride above or below the groove-wall wear zone.

9. Fix Record Tracking Problems

If the record is severely warped and is having trouble tracking, try the coin trick that your grandparents used. Neutralizing the counterweight of the tone-arm, and applying some mass in the form of coins on the tone arm shell works wonders for warped, skipping discs. There is a "physics" basis for this technique. Ask your grandparents, and they will explain it to you. If they are not available, the technique involves the "second moment of inertia". The rear counterweight has little variable effect on this parameter. So, placing coins on the cartridge shell dramatically decreases the time constant of the second moment of inertia, allowing the tone-arm system to better track warped records without launching the tone arm into deep space. Pennies are used for mild cases, dimes are used for slightly tougher cases, nickels for more serious cases, and quarters are used for basket cases! Do not worry too much if a basket case causes the pickup cartridge to "bottom out" on the record producing severe thumps in your stereo reproduction system. Most of these thumping artifacts can be removed later with the DC8/DC FORENSICS tool-set.

10. If your record skips, try Half Speed Re-Mastering

If your record still skips, try using the half-speed re-mastering process. To perform half-speed mastering, you will need a turntable with wide-ranging speed variability (some turntables with pitch controls may have enough variability to hit some of the half-speed RPM values that you will require), and a reel-to-reel tape deck with at least two speeds. Set the turntable to one-half the record RPM rating. For example, if your record is a 78 (78.26), set the turntable for 39.13 RPM. Set the tape deck speed for the speed just under the highest speed available. For example, if your tape deck has 15, 7.5, and 3.25 ips settings, use 7.5 ips to transfer the record to tape (a three minute song will take 6 minutes for the transfer).* To restore the original pitch of the recording, transfer the tape onto your hard drive at 15 ips. Another method involves the use of the Change Speed filter. Transfer the record onto your hard drive with the turntable running at 45 RPM. Then, using the Change Speed filter, correct the pitch according to the following list:

1. 78.2 RPM record - Use +73.7% pitch increase (flat line contour)
2. 78.8 RPM record - Use +75.1% pitch increase (flat line contour)

3. 80 RPM record - Use +77.7% pitch increase (flat line contour)

When this procedure is utilized, equalization gets a bit messy. For example, if the correct turnover frequency was supposed to be 500 Hz, the setting on your pre-amplifier must be re-adjusted to 250 Hz to compensate for the fact that you are reproducing the disc at half its intended speed. If you are not using exactly half speed as your re-mastering speed, then you must use ratio-proportioning to determine the correct turnover frequency setting for your pre-amplifier. "Coin therapy" and/or half speed mastering should solve most warped record transfer problems. If these do not work, consider these two alternatives. The first involves utilizing a microscope to view the stubborn portions of the record. Look at the problem groove under a microscope with 50X, and with a very sharp tipped hobby knife; clear a pathway for the stylus to follow during reproduction. Minimize, but do not become overly concerned with groove damage because the software program can compensate for this factor to a large degree later. Remember that if you are dealing with an acoustical recording, equalization speed compensation will not matter because you should be transferring with a flat pre-amplifier response, no matter what the value of re-mastering speed that you use.

* Warning! Listening to the Blues half-speed can be very, v e r y , d e p r e s s i n g .

11. Adjust the Gain and Balance

Adjust the gain and balance of the audio input signal feeding your sound card. Play a portion of the song which you believe has the loudest crescendo or passage, and make sure that the system does not overload on any of the musical transients as indicated on the Level Meters. On the other hand, make sure that the gain is not so low that you are recording "in the mud." If you do, you will lose signal resolution and the signal-to-noise ratio will not be optimal.

Important Note:

Occasional overloading due to severe clicks and/or pops is allowable and will not adversely affect the sound restoration process. However, it is imperative to be sure that any overloading is due to noise transients, and not due to musical transients such as drum "rim shots," etc.

12. Choose the Appropriate File Conversion Technique

Choose the File Conversion that is most appropriate for your .wav file, if necessary. In a few cases this procedure is not necessary. For example, if you are starting with a monophonic .wav file, there is no need for a file conversion. Also, if you are starting with a stereo .wav file and intend to maintain it as a stereo .wav file, no file conversion will be necessary. However, in many cases, you will be dealing with a stereo-recorded .wav file of a monophonic source that must be converted to the cleanest format for further processing. This is the case with most 78-RPM laterals and some vertical (hill and dale) transfers wherein the A-B function was unable to be performed by your pre-amplifier. The method for determining the best file conversion for a 78-RPM lateral transfer is something you will have to subjectively judge for yourself. Use the preview function in the File Conversion feature found under "Filter". This will allow you to quickly hear the results of your selection.

First, try to listen to the material in stereo. This will be your baseline for comparison (reference). Next, listen to a file conversion from Stereo to Mono (Left Only). Compare the results of this audition to a file conversion to Mono (Right Only). This is essentially allowing you to compare the effects of the wear on the inner versus the outer groove wall of your recording. In many cases, these two will sound the same. However, in some cases with extremely worn recordings, one groove wall will sound much cleaner compared to the other. If this is the case, make a note of which of the two sounded cleaner (containing the least distortion).

The next comparison will be between the best of the two groove walls and the Mono (L+R) file conversion. In most cases, Mono (L+R) will be the cleanest version of all of the alternatives. This is due to the cancellation effect of the record vertical displacement component that contains record and turntable rumble when using this file conversion.

So listen carefully to the "bottom end" (bass) differences between the various conversion alternatives. Also, very often, some of the clicks and pops will diminish in intensity with the Mono (L+R) conversion. If you have transferred a vertical recording using a stereo cartridge, the

only file conversion that generally makes any sense is the Mono (L-R) feature. This rejects the entire lateral component of noise signal from the transfer, preserving only the important vertical vector. After these decisions have been made, create a new file using the appropriate file conversion algorithm to be used in the next step in the restoration process.

Important Note: When using the Stereo to Mono file conversion, it is imperative to verify that both gain controls are set to -6 dB in order to assure that you do not overload the Destination channel input. -6 dB is the DC8/DC FORENSICS default setting for this feature.

13. Filter out Residual Rumble

Filter out any residual "rumble" left on your Source File, by implementing the High-pass Filter. Cutting all frequencies below 30 to 50 Hz with an 18 dB / Octave slope can be useful for this purpose. The Diamond Cut Virtual Pre Amp (VPA) includes a 30 Hz Rumble feature making this process simple. If you are dealing with acoustical material, you will probably want to set the frequency of the filter even higher to around 100 Hz since very little information generally exists below that frequency. For this, you will have to use the High Pass Filter because the VPA rumble filter is fixed at 30 Hz. Don't be afraid to experiment and use "Preview" mode to help choose the best value for your material.

14. De-click using one or more of the Impulse Filter(s)

First, remove Big Clicks due to cracks in the record with the Big Click Filter. Next proceed to De-click the record using the Expert or the EZ Impulse Filter. Usually, the EZ Impulse filter will perform this function just fine. If your record is very "hissy" (lots of top-end noise as will be found in many early 78s) then it is sometimes useful to run a low-pass filter first set for around 14 kHz and 6 dB/Octave. The reason for this is that a lot of "hiss" can fool the algorithm into thinking that there are a lot of sibilant sounds on the recording, which will move the algorithm's threshold too far up to capture ticks and pops effectively. This filter should never be run at a frequency lower than 12 kHz at no steeper than 6db/Octave. (This rule never applies to vinyl LPs or 45s.) See the section on Low-Pass Filtering for details on its operation. If your record is not "hissy" or once your "hissy" record has been Low-pass filtered, it is now ready to be de-clicked by an Impulse Filter. If you

are using the Expert Impulse Filter, for 78-rpm records, a good initial setting for the "Size" control is 5. Make sure that the filter is not in "Vinyl LP" mode. Set the Tracking adjustment all the way down (to a setting of 1) and perform all of your adjustments with the threshold control. Adjust the threshold until the clicks are removed, but the process does not distort the sound signal. Remember to utilize the preview feature to establish the best settings for your particular restoration job. Only in rare circumstances, as may be found on a very "high fidelity" 78 rpm record, will you have to employ the services provided by the tracking control, to move filter threshold out of the way on high-frequency musical transients. A good alternative to the Expert Impulse Filter that usually does a reasonable job is the EZ Impulse Filter.

After running one of the Impulse Filters, listen to the entire recording and drop markers anywhere you find any clicks, pops, or thumps that the Impulse Filter did not eliminate. Try selective De-clicking utilizing "Sync Mode" (found in the "View Menu") and the Impulse filter or use one of the manual de-clicking processes to eliminate these. You can choose between the manual de-clicking process using copy and paste over, or you can use a much simpler process utilizing the interpolate feature. In extremely rare cases, it may be necessary to use the "Cut" feature when the anomaly is extremely long in time duration.

Some of these techniques may not re-insert a signal as close to the original signal as does the Impulse Filter, so manual de-clicking should only be performed as a last resort. If there are a high number of impulse events that were not eliminated by the Impulse Filter, you should consider re-adjusting the filter's parameters and running it again before attempting to manually de-click the remaining noise artifacts. Multiple passes through the impulse filter with different settings for each pass is often useful.

When choosing a manual de-clicking technique, consider the following tradeoffs:

- The "Copy and Paste Over" method provides some attempt to re-insert a signal in the location of the transient, but it takes more operations to perform a single de-click with this method.

- The "Mute" method requires less steps to perform its function, but it replaces an impulse event with silence. This is OK when the clicks are very short in time duration. Your ear will integrate out the silence during the muted sector of the .wav file. However, longer events, when de-clicked with the Mute function, will produce some inter-modulation distortion that may be noticeable.
- The "Interpolate" key is your best alternative in terms of ease of use and accuracy, so long as the problem is limited to a relatively short interval of time. The program will prompt you if you have chosen too large an area to be interpolated.

15. De-Crackle the Recording

Many 78s have clean surfaces, and do not require this step in the restoration process. However, some very worn records, or records that were stamped utilizing low quality resins and fillers benefit from this step. So use the Narrow Crackle or the Median Filter to reduce "crackle" on recordings that need it. Crackle is essentially low amplitude clicks and which occur much more frequently. (It sort of sounds like "Rice Krispies".) It is important to perform this step only after the initial "de-clicking" step has been performed, and before the de-hissing step is performed, if the best results are to be obtained. Usually, "Size" (in the case of the Narrow Crackle Filter) or "Sample" (in the case of the Median Filter) settings from 3 to 6 produce the best results. Using too many samples will create a distortion (sort of an "edge"), which is particularly pronounced on vocals. But experimentation is the only way to determine the most appropriate "Size" or "Samples", since material varies widely in crackle content. Use the "Preview" feature to aid in the selection of the most appropriate setting.

Important Note:

Very early cylinder recordings may benefit from the application of the "Average" filter for de-crackling. Start with a small number of "Samples" and work your way up until you have achieved a good result. If you are not sure whether to use the Median or the Average filter, try both and decide which one works the best based on your own specific observations.

16. De-hiss the Recording

De-Hiss the recording using one of two methods. Even if you have a relatively quiet record, there will still be some residual hiss that you may wish to reduce. There are two alternatives to choose between. The Continuous Noise filter is the most effective at eliminating wide band noise all the way down to the bottom-end of the audio spectrum. Consider using the Artifact Suppression mode when performing this step. For details, please refer to the Filter section describing the Continuous Noise Filter.

However, it has the possibility of introducing some digital artifacts onto your .wav file if it is not set correctly, particularly with high settings of the attenuator control. Start with a setting around 20 dB. If you set this control too high, it will also reduce some of the "ambiance" of the recording with the tradeoff of providing a great deal of noise reduction.

Your other option is to use the Dynamic Noise Reduction filter. It will not produce as much noise reduction as the Continuous Noise filter, but it is easier to set up without introducing digital artifacts onto your Destination .wav file. So the choice will have a lot to do with the condition of your Source File, and your own taste with regard to the tradeoffs that were just outlined. Trial and error is a good method for sorting this one out.

17. Eliminate Line Frequency Hum and/or Buzz

Eliminate 50 or 60 Hz hum and their harmonics from your recording utilizing the Notch or Harmonic Reject filter. When "notching" out hum, be sure to set the bandwidth control to the smallest value that is effective in order to minimize any effects on adjacent frequency information. Similarly, when attenuating a "Buzz" with the Harmonic Reject Filter, minimize the number of harmonics required to do the job at hand. Some early recording had some hum induced by the amplification system used to cut the master. Two frequencies will sometimes be heard. One is often the line frequency fundamental of either 60 Hz on American made records, and the other is 50 Hz on European makes. Another frequency sometimes heard is at twice the

line frequency and is usually due to faulty filter components in the power supply section of the master cutting-head amplifier. If you hear any of these, they can be greatly attenuated with the Notch filter. This filter can be set to have a very narrow bandwidth, and so will have very little sonic effect on the adjacent frequencies. Remember, if your recording should contain "Buzz" rather than "Hum", use the Harmonic reject filter instead. You will learn through experience to discern the difference based on the sound that these two anomalies produce. And, usually, when a recording contains a lot of "Buzz" that usually means that your record transfer setup has a technical problem and the "Buzz" is not on the master. It is always best to clean up your record transfer setup rather than trying to fix this kind of problem with the software post facto.

18. Provide a Fade-In and a Fade-Out Sequence

Provide a smooth Fade-in and Fade-out sequence using the features with the same name. DC8/DC FORENSICS allows you to choose between a linear and a logarithmic gain vs. time curve. Experiment to determine which curve is the best for the material you are dealing with. After you are done setting up your fade-in and fade-out process, use the "mute" function to eliminate any extraneous noises or signals that precede the fade-in and succeed the fade-out sequence.

19. Add Your Own Personal Touch to the Transfer

DC8/DC FORENSICS provides you with a 10 band Graphic Equalizer and a Paragraphic Equalizer that you can use to create a more pleasing tonal balance to your audio restoration. It can be found in the Filter Menu. Alternatively, as you transfer the restored .wav file back onto your audio media from the computer sound card, you can consider feeding the signal through various analog signal-processing devices to give it your own personal flare. Software systems such as parametric equalizers allow you to bring out sounds that you consider to be too understated, and you can attenuate sounds that you may believe to be overemphasized

Some people like to add some ambiance to the transfer with a little bit of reverberation, which can be accomplished with the Reverb tool, or an external reverb. Devices and effects such as Spectral Enhancers and Harmonic exciters (found within the VVA effect) can provide

interesting effects especially on vocals. If your recording is missing too much of the "top-end" you can enhance it with the Virtual Valve Amplifier exciter. It will synthesize harmonics of the upper registers of the musical scale, and allow them to be mixed back into your original source material. If your material is exceptionally noisy, consider using the "High Noise" mode when applying the VVA to old 78s. Another interesting effect to consider is the Punch and Crunch multiband Dynamics processor. This can be used to restore some of the original dynamics which may have existed during the recording session.

If you are performing a sound restoration for commercial release, it is important to provide a product that will sound good on most audio systems to most people. For example, if you have a very low quality reproduction system, you may be providing equalization, which is really compensating for the fact that your sound system has poor frequency response, rather than compensating for a lack of certain frequencies on the recording itself. On the other hand, if your system is the state-of-the-art, you may want to listen to your EQ on a cheaper sound system as well, to be sure that the recording does not "break up" due to excessive bass. With any reproduction system (and sound room), it is a good idea to "flatten it out" before making subjective decisions regarding your final EQ. Use a pink-noise generator and a Real Time (spectrum) Analyzer (RTA) as the stimulus / response system for the measurement. Connect a graphic equalizer (preferably 1 / 3 octave, 30 band) in cascade with the output of your reproduction system pre-amplifier. Flatten out the response of the system / room utilizing the graphic equalizer, moving the measurement microphone to various locations in your listening area. Flatten out each channel independently. Lastly, when making your final EQ decisions, get someone else's opinion as well as your own (and include both sexes). Since audio "quality" is a highly debated and subjective area of discussion, some "opinion averaging" is in order when engaging in commercial releases. If you do not seek the opinions of others at this phase, you may find that your customers will volunteer it to you, and at a point where it will cost a fortune to fix! (If the reviews have already gone out, it may be too late to fix it at all!)

When you are done with the restoration process, you will have to transfer the contents out of your computer hard drive and into some other format. The most obvious method simply involves the use of the

D-A converter in your sound card. However, this method will produce a poorer transfer compared to direct digital, since it will introduce a small amount of distortion and noise (hum) onto the recording. If the audio restoration you are performing is for commercial release, or you are simply interested in obtaining the highest possible transfer quality back to another medium, you should consider using a digital-to-digital transfer. There will be no generational losses if this technique is used. Another alternative for achieving a loss-less transfer of your Destination .wav files would be to use a CDR drive. These units are readily available, low cost, and can be mounted directly in one of your computer bays.

Restoring a Recorded Telephone Conversation

Recorded telephone conversations sometimes are very difficult to decipher due to gain variations of "out-of-spectrum noise", "in-spectrum-noise", poor frequency response and high levels of distortion. This is particularly true of sound transfers made from Digital Communications Recording Systems. The following five-step procedure is useful for restoring such recordings. It is recommended that these steps be performed in the order outlined below. The process is arranged in an order that will require only the poorest recordings to require all of the steps outlined. Some recordings will require only one or two steps for adequate intelligibility. Keep in mind that you are not trying to tune for high fidelity here; improved intelligibility of the conversation is the only goal of this procedure.

1. If there is a variation in the gain (loudness) of the recording depending on which party is talking, or where you are in the conversation, use the DC8/DC FORENSICS Gain Riding Procedure first in order to even out the levels.
2. Next, apply the Band-pass filter. This filter is used to remove "out-of-spectrum-noise" from the signal. Use one of the two Speech Filter settings indicated below, choosing the one that is most effective for the particular material you are dealing with:

Standard Speech Filter:

- Low Frequency - 300 Hz
- High Frequency - 3,000 Hz
- Slope - 12 dB / Octave

Steep Slope Speech Filter:

- Low Frequency - 250 Hz
- High Frequency - 3,500 Hz
- Slope - 18 dB / Octave

If none of these setting improves the signal-to-noise ratio of the signal, experiment with your own values for the Band-pass filter.

3. To reduce "in-spectrum-noise" next apply the Continuous Noise Filter. Before running the filter, highlight a sector of the conversation containing a slight pause between the two parties talking for use as the "sampled noise" baseline for the Continuous noise filter threshold setting. This sector should contain only background noise.
4. If the signal remains "muffled" or "garbled", try applying the Median filter. Start with the Samples control set to around 9, and increase to as high as 19 or more until the consonant sounds and the sibilant sounds become more pronounced and intelligible.
5. Lastly, try applying the Graphic Equalizer to improve the overall frequency response of the conversation.*

***Note:** If there still remains a large variation in loudness between the two speaking parties, apply the ALC feature found in either the Dynamic Processor or the Punch & Crunch Effect.

Rumble Reduction

Rumble is a low frequency random noise found on record recordings. Generally, the noise in the spectrum below 20 Hz is considered rumble. However, with very old 78 and 80-rpm recordings, rumble can often be heard with higher frequency content. DC8/DC FORENSICS provides you with three possible methods for substantially reducing rumble on a recording. But first, it is of course important to perform your transfers with a low rumble turntable to begin with. There is no sense in making the problem worse by using a cheap turntable (especially the types which use a thrust bearing which is of the ball race variety). Turntables whose platters are thrust supported on a single ball (or point) will offer the lowest degree of turntable rumble.

Method #1:

If you are transferring a lateral 78, or a lateral monophonic LP or Monophonic 45 RPM record, transfer it to your hard drive via a stereo cartridge. Then, using the File Conversions feature, convert the file from "Stereo" to "Mono L+R." Since most of the rumble on these types of records are contained in the vertical vector only, the Mono L+R algorithm will cancel out this noise signal, and only preserve the lateral (horizontal) component, which does not contain rumble.

Important Note:

This technique will not work on Hill and Dale or Pathe´ (groove width modulated) or stereophonic recordings.

Method #2:

Use the Continuous Noise Filter as prescribed in an earlier tutorial. It is only important to realize that the signals you are dealing with will be those at about 60 Hz on down. So when you experiment with the threshold line position, only adjust it in the lowest octave to eliminate rumble. When adjusted correctly, you should be able to preserve the last octave of the audio band in terms of actual signal, while rejecting the lower signal level rumble.

Note: This technique will work on all types of recordings, not just monophonic lateral recordings.

Method #3:

Use the High-pass Filter, with settings of 60 Hz on down, and the slope control set to 18 dB / Octave. Experiment until you achieve the desired results using preview mode.

Note: This technique will also work on all types of recordings, not just monophonic lateral recordings. However, it is not as good as method number 2 because it will also eliminate some of the low bass of your recording as well as its rumble content.

Method #4:

Use the Channel Blender Effect with the "Vinyl LP Cancellation Filter" or the "Vinyl LP Bass Clarifier" preset enabled. Often, this method produces the optimum rumble reduction results on vinyl stereo recordings.

Simulate Stereo from a Mono Source – Method #1

1. Open the Monophonic File to which you desire to add "Stereo Simulation."
2. Click on "File Conversions", found under the Filter Menu.
3. In the "Source File" box, click on "Mono."
4. In the "Destination File" box, click on "Stereo."
5. Click on "Run Filter."
6. When the conversion process is completed, close the File Conversions dialog box
7. Under the File Menu, click on "Make Destination the Source."
8. When the "Save As" dialog box appears, click on "OK" if you are satisfied with the automatically assigned temp file name.
9. Under the Effects Menu, click on "Reverb."
10. Using the "Preview" mode button, listen to the effect that the reverb is having on the file. Adjust the various reverb parameters until you achieve the desired stereo effect.
11. When you are satisfied with the reverb settings, click on "Run."

Simulate Stereo from a Mono Source – Method #2

1. Open the Monophonic File to which you desire to add "Stereo Simulation."
2. Click on "File Conversions" found under the Filter Menu.
3. In the "Source File" box, click on "Mono."
4. In the "Destination File" box, click on "Stereo."
5. Click on "Run Filter".
6. When the conversion process is completed, close the File Conversions dialog box.
7. Under the File Menu, click on "Make Destination the Source."
8. When the "Save As" dialog box appears, click on "OK" if you are satisfied with the automatically assigned temp file name.
9. Under the Filter/EQ Menu, open the Paragraphic EQ.
10. Activate the Left (L) channel by clicking on the "L" button on the Toolbar (on the top of the Diamond Cut software program).
11. Find the Paragraphic EQ preset called Stereo Simulator Left Channel Comb Filter* and then click on "Run" in the Paragraphic EQ dialog box.
12. Make the Destination the Source.
13. Activate the Right "R" channel by clicking on the "R" button on

the Toolbar and then click on “Run” in the Paragraphic EQ dialog box.

14. Make the Destination the Source.
15. Activate the Right ® channel by clicking on the “R” button on the toolbar.
16. Find the Paragraphic EQ preset called Stereo Simulator Right Channel Comb Filter* and then click on “Run” in the Paragraphic EQ dialog box.
17. Close the Paragraphic EQ and open up the File Conversion Filter.
18. Click on the “LR” button on the Toolbar.
19. Set up the File Conversion Filter for “From Stereo to Stereo” mode.
20. Set the Time Offset for 10 mSec and Preview the filter. Adjust the Time offset while previewing until you obtain the stereo effect that you like and then Run the filter.

*Note 1: Presets with a (W) suffix will provide a stronger stereo effect.

Note 2: Sometimes, you may want to combine methods 1 and 2 for the strongest stereo effect.

Manual Splitting & Recombining a Stereo Wave file

Though we have recently implemented a new automated Split and Recombine tool, some users may prefer performing this process manually.

1. Bring the desired stereo file up into the Source Workspace.
2. Under the Filter menu, select the File Conversions feature.
3. Click on the "From Stereo to Stereo" function.
4. Lower the Left Amplitude control to -96 dB (control all the way down).
5. Run the Filter
6. Make the Destination the Source and then rename the file. (It may be a good idea to add an extension to the file name denoting that it is the right channel only.)
7. Double click on the Icon just to the left of the File Menu selector to return you to the window that contains the original stereo .wav file located in the source workspace.
8. Double click on the Source Workspace.

9. Lower the Right Amplitude control to -96 dB (control all the way down).
10. Run the Filter again.
11. Make the Destination the Source again renaming the file. (This time you may consider adding an extension to the file name denoting that it is the left channel only.)
12. Perform whatever independent signal processing you desire on each of the two files (channels). If only one channel requires processing, just process that particular .wav file.
13. When you are done with the independent channel processing, bring up the final version of the right channel into the source window.
14. Under the Edit menu, click on "Copy."
15. Bring the final form of the left channel into the source window.
16. Under the Edit menu, click on Paste and then Paste Mix.
17. Click on OK - the resultant file in the source workspace is the re-combined stereo .wav file.

Using DC8/DC FORENSICS as an Audio Waveform Analyzer

DC8/DC FORENSICS can be used as a waveform analyzer. It is capable of both time and frequency domain analysis. This waveform information may be useful to know when trying to determine what useful frequency range is present in a .wav file. It also will allow you to measure resonance or acoustic feedback, and periodic noise sources such as 50 or 60 Hz Hum (or harmonics thereof). Once this information is known, it will be easier to set up the parameters for the Notch, High-pass, Low-pass or Band-pass Filters, so that you can retain useful portions of the spectrum, while rejecting unwanted noise signals. This information is also useful in Forensic applications for measuring the time between gunshots or other events of critical importance. The waveforms, either in the time or frequency domain, can be printed out using the Print Screen procedure.

Method #1

Using the time scale in the workspace to determine a signal's frequency:

1. Zoom-In on the portion of the .wav file containing the suspect noise signal, to the extent that you can make out the peaks and valleys of the waveform of interest.
2. Note the total time duration of the "frame" which you are looking at. It is located on the bottom of the DC8/DC FORENSICS window in the status bar. (This time is the difference between the frame "end" time which is indicated in the upper right hand corner of the workspace and the frame "beginning" time that is indicated in the upper left hand corner of the workspace.) This number will be in fractions of a second.
3. You will notice that the x-axis of the workspace is divided into 10 grids. Calculate the time duration of one grid by dividing the total frame time duration by 10.
4. Count the number of cycles (cyclic peaks, or cyclic valleys), which occur within one of the ten x-axis grids. Divide the total time duration of one grid (as determined in step #3) by the number of cycles that you have counted. This is the value of the time duration for one cycle of the waveform you are observing (Tcycle).
5. Take the inverse of that time in seconds. This is the value of the frequency of the waveform you are observing ($F = 1 / T_{\text{cycle}}$).

Method #2

Using the Band-pass Filter as an Audible Wave Analyzer:

Note: This method only works well if your computer is fast enough to run the Band-pass Filter algorithms in real time in Preview Mode. Otherwise, the system will "stutter" making it difficult to interpret the results. Also, it is important to realize the subjective nature of this method of analysis. Your own ability to hear frequencies at the extreme ends of the audio spectrum is critical. You need to consider that most men begin to lose their high frequency response after the age of around 21, and most women begin to lose their high frequency response after the age of around 30 (although there are always exceptions). You may want to engage the opinion of your kids (whose hearing is generally exceptional, believe it or not) when performing

these subjective tests. You will also need a very high quality sound system for these measurements. It is very useful to have one that utilizes a sub-woofer for making the low-end measurements.

1. You will be using the Band-pass Filter in "Preview" Mode.
2. Select a Slope value of 18 dB / Octave.
3. Set the High Frequency slider control all the way up to its highest frequency.
4. Place the Low Frequency control to around 7 kHz.
5. Listen to the Material in Preview mode through a high quality audio system.
6. If you can make out any audio signals containing information and not just noise, increase the Low Frequency setting in 1 kHz increments.
7. Repeat step 5 and step 6 until all you hear is noise. This will be your upper cutoff frequency value for the material you are dealing with. (Be careful in this evaluation because some very subtle sibilant sounds add a great deal of character to an audio source. So make sure you are only hearing noise when you decide that the frequency control has been set appropriately.)
8. Next, set the Low Frequency slider control all the way down to its lowest setting.
9. Place the High Frequency control to around 100 Hz.
10. Listen to the Material in Preview mode through your sound system.
11. If you can make out any audio signals containing information and not just noise, decrease the High Frequency setting in 10 Hz decrements.
12. Repeat step 10 and 11 until all you hear is noise (rumble). This will be your lower cut-off frequency value for the material you are dealing with. (Be careful in this evaluation, because some very small low amplitude bursts of bass in sync with the music may be very important in the overall sound character of the source.)

Method #3

Using the Audio Spectrum Analyzer

1. Using the mouse, highlight the sector of the Source .wav file that you desire to analyze. Often this will be a lead-in groove or lead-out groove of a record, in order to analyze the record background noise. However, it could be a sector of a .wav file in which

acoustic feedback occurred or Hum was observed, and you desire to measure its frequency, so that it can later be attenuated.

2. Click on the Filter Menu and select Continuous Noise.
3. Click on "Sample Noise."
4. A status window will pop-up indicating the "% Done" as the system performs its calculations
5. When the calculations have been completed, the status box will disappear, and a graph will appear.
6. The graph of interest is shown in red. The graph will plot the Amplitude expressed in dB versus the Frequency of the selected portion of the .wav file. (The blue graph is used for a different application, and is not applicable for this function.)

Method #4

1. Bring up any of the filters or effects.
2. Under the View Menu click on the Spectrum Analyzer.
3. Set the Spectrum Analyzer controls to the desired settings.
4. Run "Preview" for the particular filter, and the post-filter frequency domain signal shall be presented on the graphical display.
5. To "freeze" the display on a particular signal, click the enable button off.
6. When running a filter, you can leave the spectrum analyzer on, but it will consume some resources, slowing down the processing of the file to a small degree. So it is best to shut down the spectrum analyzer during processing.

Ogg Vorbis Lossy Compression Tag Support

Your Diamond Cut software can both play and encode files using the Ogg Vorbis format. You can also rip files from a CD into this format as well as play then via the DCTune Library. It also supports the following Ogg Vorbis tags:

TITLE: This field is the Track/Work name and it is initialized with the name of the file being used.

ALBUM: This field is related to the collection name to which the track belongs.

TRACKNUMBER: This field contains the track number of the piece if it is a part of a specific larger collection or if it is part of an album.

ARTIST: This field contains the artist generally considered responsible for the work. In popular music, this is usually the performing band or singer. For classical music it would be the composer. For an audio book it would be the author of the original text.

GENRE: A short text indication of music or track genre.

FLAC Lossless Compression Tag Support

Your Diamond Cut software can play and encode files using the FLAC (.flac) lossless file compression format. FLAC produces a reduction in file size of roughly one-half without any loss of sound quality. The “encoding / compression” control actually affects the degree of compression that the system will achieve and not the actual audio quality of the resultant file. However, higher values of “compression” will take longer for your machine to create due to the increased complexity of the math involved, but the resultant file size will be smaller. The software supports the following FLAC tags:

TITLE: This field is the Track/Work name and it is initialized with the name of the file being used.

ARTIST: This field identifies the artist or songwriters who are involved with the creation of the file.

ALBUM: This field is related to the collection name to which the file belongs.

GENRE: This field allows you to specify a type of music relating to the file.

TRACKS: (Numerical Value) This provides a field to identify the number of tracks involved in the file.

YEAR: Use this to identify the year of the file creation or the year of the audio restoration or re-mastering date.

mp3 Lossy Compression Tag Support

Your Diamond Cut software can supports the following mp3 lossy compression file tags:

NAME: Use this as a description field for the mp3 file.

ARTIST: Use this to describe the author or artist associated with the file.

GENRE: This field allows you to specify a type of music relating to the file.

TRACK: (Numerical Value) This provides a field to identify the number of tracks involved in the file.

YEAR: Use this to identify the year of the file creation or the year of the audio restoration or re-mastering date.

Section 4 – Tech Support

Trouble Shooting

This is your one stop shop for problems or questions (FAQs) that you may encounter while working with our software. This list of problems and solutions constitutes about 90% of the overall problems our techs deal with. Chances are...if you have a problem with our product, the solution is listed below. Please read this before contacting us.

This section lists common questions and problems users have with DC8/DC FORENSICS.

Q: How do I control the recording level of the audio signal?

A: In Windows, there is a speaker Icon in the lower right hand corner of the Taskbar. Double-click on this Icon to bring up the control panel for your sound card. There are level controls for the Mic or Line inputs of the sound card. For Sound Blaster type cards, double click the yellow speaker icon and then choose Options/Properties. Click on Recording, and then click OK. You'll now have your input mixer up. Also be sure that the correct input is selected (Mic, Line, or Aux) for your particular recording setup.

Some sound cards have their own audio control panels that are better suited to controlling the audio inputs. These are usually available from the Start Menu

Q: I'm trying to record and I hear the audio, but the Record meters are not jumping.

A: Most sound cards come with the Line Input not selected. This causes a "no recording" symptom. But first, make sure that the sound card is installed correctly. Just go to the Control Panel/Multimedia and make sure that the sound card is selected as the default playback and recording audio device. If it is, the sound card is most likely working okay. Next, bring up the Preference dialog box and select the "Sound Card" Tab and make sure that the inputs and outputs are set to the correct sound card. If the system still is not recording, double click on the yellow speaker icon on the tray and choose Options/Properties and click on the Recording radio button. Click OK. This brings up the input mixer. Make sure that the slider labeled "Line Input" is selected

(it usually will not be). When you do this, the meters in the Record window will start modulating immediately, indicating that the system is now working fine.

Q: I get a Windows error message when I try to record that says something about "this is beyond the capability of this device" or "The specified format is not supported". What's this?

A: This just means that your sound card is not capable of recording with the sample rate or bit width you have selected. For example, if you try to record a 24-bit file with a 16 bit sound card, you'll get this message. Most sound cards will work with 44.1 kHz Sampling rate, 16 bit, stereophonic recording settings. This could also mean that Windows is assigning .wav files to be played by some device other than a sound card...like a modem. Make sure Windows is aware that you want .wav files assigned to your sound card's output.

Q: I click on Play and don't hear anything but the VU meters are jumping fine in the program.

A: If the cursor is moving across the screen during playback and the VU meters are jumping, you can be pretty sure the audio is indeed playing. Usually, this just means that you don't have your speakers plugged into the soundcard output where the audio is appearing. Check under Edit/Preferences/Soundcard to see what the output device is set to and make sure your speakers are plugged into that set of outputs.

Q: My levels are low during recording.

A: Maybe not. You should normally not record all the way up to 0db (the red area) with digital systems. Remember, you'll likely need some "head room" in order to be able to increase volume with some of our tools such as the Paragraphic EQ. Peak levels of -4 or -6 db will give great results. If your levels are REALLY low, check to make sure you have the right type of signal going into your sound card. You cannot plug a turntable directly into a soundcard, for example - you'll need a preamp for this.

Q: DC8/DC FORENSICS contains a Fast Edit mode and the classic two-file mode. Which one should I use?

A: Beginners tend to like the Fast Edit mode since it works more like other audio editing programs. It's intuitive, simple and fast. However, as you learn the program we think you'll find the classic two-

screen mode to be the most powerful audio restoration environment. Our "Source and Destination" way of working is optimized for those situations where you want complete control of the files being edited created and saved. This is especially useful for forensic situations where an audit trail is required.

Q: I clicked Fast Edit in the Preference box and I'm still in the Classic mode.

A: You need to close the program and restart it for this change to take effect.

Q: I am getting crashes when I try to play audio. What do I do?

A: DC8/DC FORENSICS uses both the new Windows Driver Model (WDM) method of communicating with sound cards as well as the more common MME method. WDM is new to many sound card manufacturers and some of these new sound card drivers may have bugs in them. If you are getting a crash when you try to play or preview a filter, please check to see if a new sound card driver is available from the manufacturer of your card. If this doesn't do the trick, look under Edit/Preferences/Sound Cards and check the box which says "re-initialize on play". This makes the sound card return to a known state every time you play and may help you. You can also select between WDM and MME in this area to coincide with the most reliable support from your sound card.

Q: Why does Preview mode sound like it is stuttering?

A: All of the filters in DC8/DC FORENSICS require a fair amount of processing power. If your computer cannot complete the processing fast enough to keep up with the audio stream, then the preview mode will stop and start in short bursts that sound like stuttering. This effect can be reduced or eliminated by increasing the number of "Preview Buffers" in the Edit/Preferences/Soundcard dialog box. Regardless of stuttering in Preview, all "run filter" operations will be stutter free. Putting many filters in the Multi-Filter box will likely cause stuttering on even very fast machines. You may also want to check and see that you're using your sound card's newest drivers.

Q: I noticed that a Live Mode is now included in DC8/DC FORENSICS. What's this for?

A: This allows you to take a "live" audio stream and process it without first recording it to hard disk. With this function, you can make your computer an integral part of your home stereo system, ham radio station or SWL station. Imagine adding the sound of a \$15,000 vacuum tube amplifier to all your audio without having to record and process it. As an example - take your CD player and plug its outputs into your sound card input. Take the sound card output and plug it into your home stereo receiver. Now run DC8/DC FORENSICS and choose Multi-Filter. Put some enhancement modules and/or noise filters into the signal path and click on "live preview". All these tools will be applied to every CD you play in real time.

Q: I get an error when I try to use Live Preview mode. What's up?

A: Your soundcard must be able to play and record at the same time. If you have a cheap soundcard, it's time to upgrade!

Q: I have a Whackmaster 6000 sound card I bought in 1992. Is this OK?

A: Who knows? We don't remember that year at all. Just about all of the new sound cards include a new type of driver called WDM. This is Microsoft's effort to fix a lot of earlier problems with various sound card implementations. If your really old soundcard works, then you're OK. Again, you can try going to the Preferences/Sound Card area and selecting MME as your method of communication. In many cases, even the old Whackmaster is supported by this driver method.

Q: I have a 192 kHz / 24 bit sound card. Should I record using these settings?

A: Remember, your recordings will never be better than your originals. If you are recording vinyl records and are making CDs, we recommend 44.1, 16, stereo. If you have high quality masters or are making DVD audio disks, then go with the higher settings. DC8/DC FORENSICS doesn't care how you record, but higher sample rates and wider bit widths will result in larger files and more processing time.

Q: How do I record my records using a turntable?

A: A phono cartridge puts out a very tiny signal. You cannot plug a turntable directly into your soundcard since it doesn't amplify the incoming signal enough. You'll need a preamp. If you have a stereo receiver with a control labeled "phono" on the front, you're all set since

this has a preamp built in. If your receiver doesn't have this (many modern ones don't) or your receiver is too far from the computer, get a stand-alone phono preamp.

Q: How do I connect my computer into my existing stereo system?

A: On your receiver or master component, take the "tape out" or other line level output and plug it into the line level input of your soundcard. You can also take the line out of your soundcard and plug it into the "tape in" of your receiver though most folks just connect the soundcard output to their computer speakers.

Q: My computer crashes in Preview mode with the Multi-Filter.

A: If there is any weakness in your system, a Multi-Filter preview will likely find it. This can require thousands of calculations per second (or more) while audio is flowing. The audio can be presented to your sound card in a start/stop fashion as well (though the buffering on your sound card and in DC8/DC FORENSICS makes for smooth playback). Sound card designers seldom test for this environment and you can uncover bugs in the drivers. Make sure you have your latest sound card and video card drivers and try reducing the amount of video acceleration your video card is set for (this reduces the overall system overhead a bit).

Q: Will increasing the amount of RAM in my computer make DC8/DC FORENSICS run faster?

A: DC8/DC FORENSICS does not require huge amounts of RAM. If your computer system has 2Gbytes of RAM, then further increases will not appreciably speed up the program. The software uses disk-based processing so hard drive speed and raw processor speed will generally have a greater effect than increased RAM beyond a certain minimum. A faster processor WILL make a big difference in the number of simultaneous filters you can use.

Q: I want to purchase a system optimized for use with the software. What sort of system will provide me with the fastest performance of all of the various DC8/DC FORENSICS filters, effects, and editing features?

A: You should purchase a computer with the fastest clock speed in your price range. Look for hard drives with fast access times and

rotational speeds. Purchase a high performance sound card. Lastly, turn off all superfluous programs that may be running in background.

Q: How do I avoid producing dropouts during recording or playback?

A: Make sure that you have reviewed all of the following:

1. Make sure that you are using the latest drivers for your sound card. They can usually be obtained from the card manufacturers or Microsoft's web site.
2. Make sure that the screen saver and all power management functions will not kick-in during recording or playback. By default, the screen saver has a 1-minute timeout, so after 1 minute of no keyboard or mouse activity, the screen saver will kick-in. This flurry of disc activity will put a glitch on the recording or playback of .wav files.
3. Turn off all power saving features, or set their timers to a value of time greater than the longest musical selection that you want to record or play.

Q: I have a vinyl LP that is very noisy, and still has too many clicks after processing. What can I do?

A: Try running the Ez-Impulse filter or running the Expert-Impulse filter twice or more. First run it with the Tracking control set to zero, and adjust the threshold control to remove just the largest clicks. When done, make the destination the source, and re-run the filter with the Threshold set back to 1, and adjust the tracking control to get the smaller clicks. Another thing to try is to use the File Reverse feature, and then process the vinyl recording through the Impulse Noise filter. When done, re-reverse the file. Also, consider trying HQ mode, which is especially good for detecting very small clicks when adjusted correctly (small size values around 5). If you still have clicks left, it's likely because the clicks are longer than the program is allowed to automatically identify and repair. Use the Interpolate function to get rid of any remaining clicks.

Q: I have a record with one major scratch on it running from the center to the outside in a spiral pattern. How do I eliminate the loud "click" which I hear which occurs once per revolution of the record?

A: You should use the Impulse Noise filter in the following way. Set the Tracking control to its minimum value. Set the Threshold to its maximum value. Set the "samples" to about five. While in Preview mode, slowly decrease the Threshold control until you see the click counter increment once for each click, which is occurring. Do not increase the Threshold control any further than necessary. Next, increase the "Samples" control until the click is not longer audible. This technique is also useful for getting rid of the clicks produced by cracked 78-RPM shellac records. It is even possible to take a broken 78, glue it together, and then after transferring it, remove the clicks, which occur at the breakage points.

Q: I just transferred an old stereo recording to my hard drive for processing, but I am not sure whether or not the left and right channels are in their correct positions or transposed. How can I determine if they are correct?

A: If the material is contemporary pop, it is difficult to make this determination. However, if the music is orchestral in nature there are some obvious orientations to listen for. For example, the left channel should be dominated by any cymbal crashes or the tinkling sound of a triangle. Also, when Violins come in they should dominate the Left channel. Double basses should be more prominent on the right side of the soundstage. French horns should be heard behind the woodwinds, while trumpets and trombones should generally project from the right rear. If this is not the case, the channels are probably reversed and should be corrected by applying the File Conversion Filter Stereo Reverse feature.

Q: Some of my vinyl records sound more distorted as the play moves towards the end of a side. Why is this? What can be done?

A: This distortion can be due to several sources of problem. A poorly mastered lacquer could have been the culprit. Not all mastering engineers were careful to "hold down" the dynamic range as the recording progressed, creating a great deal of groove wall over-modulation. However, a more likely scenario is that the record was played with a worn stylus. Have someone who is familiar with styli check it out under a microscope to be sure that it is not worn out or damaged. If it checks out to be okay, then your vinyl recordings may have been damaged in past times as a result of being played many times with a defective stylus. One thing that can be done is to selectively

apply a Low-pass filter to the last few cuts of the recording to reduce this annoying anomaly. Also consider applying the Filter Sweeper to compensate for this problem.

Q: How do I generate a simulated stereo Wave file from a monophonic Wave file?

A: Start with a monophonic file that has been de-noised, and convert it to stereo using the File Conversion Filters. Some stereo effect may be added here by applying a little "Time Offset" during this process. Next, make the destination the source, to get a new source file. Run the Reverb effect with a Small or Medium hall, setting the decay to a low number and the early reflections level nearly to zero.

Q: Does the order in which I process noise out of a Wave file matter?

A: Yes. Always remove clicks and pops with the Impulse Noise filter before de-hissing a recording using either the Continuous Noise filter or the Dynamic Noise filter. Never reduce the bandwidth of an audio signal before applying the Impulse Noise filter (and this applies both in the analog side of your signal path as well as the digital side.)

Q: I have an analog tape recording with clipping distortion due to over-modulation during the recording process. Is it possible to "soften" the clipping sound in order to reduce the harshness produced during the overloads?

A: Clipping distortion can usually be substantially reduced by applying the De-clipping filter. This is the preferred method. However, clipping distortion can sometimes also be reduced by utilizing the Impulse Noise filter. If the clipping distortion occurs at the peaks of the waveform, set the tracking control set to its minimum value, and the threshold set to maximum. Highlight a segment of the recording that contains distorted and non-distorted material. In preview mode and with accuracy optimization checked, adjust the Threshold control until the clipping distortion is reduced. In some cases, it may become necessary to run the Reverse NAB curve before following the above procedure. After the distortion has been reduced with the impulse filter, it will be necessary to run the NAB curve to re-correct the recordings equalization. These two curves can be found in the factory preset listing under the Paragraphic Equalizer filter. The reason for the above two steps is that the saturation overload occurs at the tape-to-tape

head interface. The resultant overload is then phase shifted during playback by the NAB equalization circuit in the tape recorder. Utilizing the Reverse NAB curve places the clipping distortion closer to the peak of the waveform, where is actually occurred.

Q: I enjoy removing "clicks" and "thuds" manually from my recordings using the Interpolate feature. Sometimes these signals are so small that, although I can hear them, I can't see them in the waveform. What can I do to make them more visible?

A: In order to improve the view of these signals, first put the system in Sync mode which is found under the View menu. In order to see the "clicks" more clearly, run the High Pass filter with it set to 6 dB / Octave and 10 kHz on your .wav file. The "clicks" will now be more visible in the Destination window and they will be time aligned with the Source Window. Similarly, if you want to view "thuds", run the Low Pass filter with it set to 6 dB / Octave and 100 Hz on your .wav file. As with the clicks, the "thuds" will be more visible in the Destination Window.

Q: I've been using the product for extensive editing, it has always worked perfectly, but now I'm experiencing lockups and shutdowns. What's the deal?

A: Obviously, these problems are not common with this product, but one area you may want to check is your Temp files. DC8/DC FORENSICS has a maximum Temp file limit of 999, so if you've been doing extensive editing in Fast Edit Mode, you may have hit that limit and need to clean those temps before it will work properly.

Q: How do I determine which version of Direct X that I am running on my system?

A: Goto the windows "Start" button and click on Run. In the "Open" field, type in "dxDiag" and then click OK. A lot of data will be presented, some of which includes the version of Direct X that you are using.

Q: The highlighted display and play cursor appear to be visually "jittery" when editing. How to I eliminate this visual effect?

A: You probably have a second instance of your Diamond Cut software open which is competing for resources. Close the second (minimized) instance and the "jitters" will be alleviated.

Q: Where do I find my exact User Name, Serial Number and Registration Code in an earlier version of my Diamond Cut Software so that I can perform an upgrade to a newer version?

A: These data are found under the “Help About” DCArt menu item of your older software program.

Q: What are the various file paths used by the Diamond Cut Forensics version 8 Software?

A: Starting with DC Forensics8, all of the user data is consolidated into a directory called DCForensic8 in the users “Documents” or “My Documents” directory. The file paths under this directory are as follows:

DC Tunes database:	\\DCForensics8\\
Presets:	\\DCForensics8\\Presets
Example Wavefiles:	\\DCForensics8\\Wavefiles
TempFiles:	\\DCForensics8\\Tempfiles

To get to these Files, use the following paths:

Vista and Windows 7

<user>\\Documents\\DCForensics8

Windows XP

<user>\\MyDocuments\\DCForensics8

Forensics version 8 also provides shortcuts which are located in the user’s program menu to access these files.

Reporting a Problem

Your software distributor has operators available during most days during normal business hours. They provide technical support for registered users only. When you contact the distributor from which you purchased your software, please be prepared to provide the following information:

- Sound Card type
- Version of the software (Found Under Help/About Software)
- Version of Windows
- Being specific as possible, outline the problem, how it occurs and its frequency
- Version of Direct X being used
- Your User Name (as entered into the software at installation)
- Your Software Serial Number
- Your Software Registration Code Number

Most distributors prefer email for first contact, in that it gives their techs a chance to study the problem before they provide you with an answer, rather than experimentation while you're waiting on the phone. Our distributors normally have an answer for you within the next business day. If you need to contact the factory directly, here is the contact information:

Contact Information

Diamond Cut Productions, Inc.
P.O. Box 305
Hibernia, NJ
07842

www.diamondcut.com

Tel: 973-316-9111

Fax: 973-316-5098

Section 5 – Useful General Information

Glossary of Terms

Acoustical Impedance

Acoustical impedance is the total opposition provided by acoustical resistance and reactance to the flow of an alternating pressure applied to a system. More specifically, it is the complex quotient of the alternating pressure applied to a system by the resulting volume current. The unit is the acoustical Ohm.

Acoustical Reactance

Acoustical reactance is the imaginary part of the acoustical impedance. Energy is not dissipated by acoustical reactance; it is only stored there. The unit is the acoustical Ohm.

Acoustically Mastered

Acoustically mastered record recordings utilized only the energy of the sound waves created by the sound source to modulate the master cutting lathe stylus. This recording technique had none of the benefits which signal amplification can provide to the recording process. This is the method that was utilized from the time of the invention of the phonograph by Thomas Edison in 1876 up until around 1925, when vacuum tube amplifiers and microphones began to be employed in the mastering process. By 1929, all of the major record companies had switched over to the "electrical process" of record mastering.

Acoustical Resistance

Acoustical Resistance is the real term of the acoustical impedance relationship. This is the term responsible for the dissipation of energy. The unit is the acoustical Ohm.

A-D Converter

A device used to convert analog signals into digital (discrete time) signals, so that they can be signal processed by a computer algorithm. The sound card in your computer contains an A-D converter and also a D-A (Digital to Analog) converter. To be compatible with DC8/DC FORENSICS, it must have at least 16-bit resolution to realize the performance of the product. However, the software does support 8 through 24 bit resolution sound cards. In other words, your

sound card must be able to divide the amplitude of audio signals into numerically sampled representations, the smallest division being one part in 65,536 (2^{16}). 16 bit audio has the same resolution as "red book" CD Audio.

A-law Compression

This is a slight variation of Mu-law compression, which is used in Europe. For more information, please refer to Mu-law Compression. DC EIGHT supports this format.

ADPCM

(Adaptive Differential Pulse Code Modulation)

ADPCM is a data compression method in which an audio signal is quantized by the difference between the input reference signal and a prediction that has been made of that same signal. When the prediction between the actual and the predicted audio signal exhibits bits having a low variance, it is accurately quantized and fewer bits are required for digitization compared to standard PCM. Your software supports this file format.

AES

The Audio Engineering Society
60 East 42nd Street, Room 2520
New York, NY
10165-2520
USA
1-212-661-8528 / <http://www.aes.org>

AGC or ALC

Automatic Gain (AGC) or Level Control (ALC). This is a set of algorithms, which can be found in the Dynamics Processor and Punch & Crunch effects that maintains the system gain at a relatively constant value independent of input signal level. Signals below a set threshold value are upwardly expanded while signals above that threshold are compressed. This is particularly useful in forensics applications wherein the various parties who are communicating with one another are recorded at greatly varying relative levels.

AIFF

AIFF is the acronym for a PCM (Pulse Code Modulation) audio file format known as "Audio Interchange File Format." It is used primarily on the Macintosh platform. DC EIGHT and DC Forensics 8 facilitates conversions from this format to .wav.

Algorithm

An algorithm is a step-by-step procedure for solving a mathematical problem.

Ampere

(I)

The unit of electric current that is equal to one Coulomb flowing per second.

$$I = V / R,$$

wherein:

V = Voltage in Volts,

and

R = Resistance in Ohms

(also, see Ohms Law.)

Amplifier

An amplifier is an electronic system which enables an input signal to control power from a source independent of the signal and thus be capable of delivering an output that bears some relationship to, and is generally greater than the input signal. An audio amplifier performs this function producing a relatively linear relationship between the input signal and the output signal. For more information on audio amplifiers, refer to Pre-Amplifier and Power Amplifier in the Glossary section of the Help File.

Amplitude

The loudness (or intensity) of a sound at any given moment in time, which is represented on the vertical axis of the Diamond Cut time domain workspace areas. Amplitude in audio terms is usually expressed in relative terms (the ratio of two levels) in dB (decibels), although sometimes it may be represented in absolute terms such as Volts or sound pressure level.

Amplitude-Frequency Distortion

Amplitude-Frequency distortion is a deviation from a perfectly flat frequency

response over a range of interest (usually 20 Hz to 20 kHz).

Analog

An electronic system in which signals are represented, amplified, and processed utilizing continuous Voltages and/or currents (whose value could be expressed as an irrational number at any point in time) which are not quantized. DC8/DC FORENSICS utilizes several digital simulations of analog systems in its algorithms.

Aromatic

Solvents that are made up of cyclic hydrocarbons or their derivatives that, after they evaporate, tend to leave little or no residue on the surface on which they had been applied.

Attenuate

Attenuation is the process of signal reduction, which is the opposite of the process of signal amplification. Most filters attenuate signals outside of their pass-band and feed signals through with no attenuation (or amplification) within their pass-band. Some filters, such as parametric and graphic equalizers are configured to provide either amplification or attenuation at any given frequency. Devices such as volume control potentiometers, "L" Pads, "T" Pads and "H" Pads are used to attenuate signals independent of frequency, i.e. flat. "L" Pads hold either the input or the output impedance constant as the attenuation factor is modified. "T" Pads hold both the input and the output impedance constant as the attenuation factor is changed. "H" Pads perform the same function as "T" Pads, only for balanced line systems. See a table of resistance multipliers for a symmetrical (equal input and output impedance) "T" Pad attenuator in our *Charts, Graphs and Other Useful Info* Section.

Audio Frequency Spectrum

The range of frequencies between 20 Hz and 20 kHz. A very high quality audio system capable of reproducing this frequency range should be able to do so within +/- 3 dB.

Azimuth

When analog magnetic tapes are recorded or reproduced, the gap of the respective head (recording or playback) should ideally be

perfectly normal (perpendicular) to the direction of the tape movement. If, in either of the two mentioned processes, the respective head gap is off-normal (off-azimuth,) two types of signal degradation will occur. The first phenomenon results in the loss of the high-end of the audio spectrum frequency response. The second effect produces a phase shifting of one channel with respect to the other, thereby "smearing" the stereophonic image. A similar phenomenon occurs when a monophonic half-track reel-to-reel tape is reproduced on a quarter track machine. Azimuth problems can be corrected by utilizing the "Time Offset" feature found in the File Converter, which is under the Filter Menu.

Band-pass Filter

A Band-pass filter allows only a range of frequencies to be passed from its input to its output without attenuation. A wide Band-pass filter is one in which an upper and a lower corner frequency need to be defined, and often several octaves will be passed in between without attenuation. A narrow Band-pass filter is one in which only a center frequency needs to be defined, and often has a bandwidth of an octave or less. The center frequency for a narrow Band-pass filter is sometimes referred to as its resonant frequency. There are different Band-pass shapes that can also be defined for narrow Band-pass filters.

Band-stop Filter

A Band-stop filter provides the inverse response of a Band-pass filter. You can find a Band-stop filter within the Brick Wall filter suite found under the Forensics menu of your software.

Berliner, Emile

Emile Berliner is widely known as the one who commercialized the lateral cut disc record format. He first introduced his products into the market place in 1895, although he had spent the previous 10-year period developing his product. However, Emile Berliner was not the inventor of the disc format or the lateral cut method for creating the undulations on a surface. These principles were outlined in the earlier Edison phonograph patents.

Blast

"Blast" is a term that is used to describe a passage of sound on a recording which is disproportionately louder than the rest of the recording. "Blasts" can be created by poor instrument placement on acoustic recordings, poor mixes on electrical recordings, or by poor planning of microphone placement in live recordings. The term "blast" was used by recording engineers at least as early as the 1920's.

Blue Amberol

The third (and final) generation of cylinder record which the Edison Company commercialized which was about 4 minutes in length. These records were made of a celluloid recording surfaced mounted on a plaster of Paris core. They were an improvement on the Edison Gold Molded black wax cylinders, which were only two minutes in length. The rotational speed for Blue Amberols (and black wax cylinders) is 160 RPM.

Broadcast Wave (.wav) Format (BWF)

Broadcast Wave or BWF (.wav) is an audio file format that facilitates the inclusion of certain metadata within a .wav file header. It adds a "Broadcast Audio Extension" chunk to the basic Microsoft .wav format. It is a European standard per EBU document Tech 3285, version 1 (July 2001). BWF provides a seamless exchange of audio data between differing broadcast environments and equipment based on varying computer platforms. This format is supported by your Diamond Cut software program. You can create BWFs via the Save As feature found in the File Menu. You can edit the BWF header information via a dialog box within the Edit Menu. Complete information regarding this format is beyond the scope of this document and can be found on the internet.

Brown Noise

Brown noise is a form of random noise exhibiting a Gaussian distribution, which mimics Brownian motion. Its power spectrum is proportional to $1 / f^2$, exhibiting a - 6dB / Octave slope. Your Diamond Cut software can produce Brown noise in two steps. First, create a random noise file using the Make Waves Generator. Then, bring up the LIVE/Multi-Filter and apply the preset called "White to Brown Noise Converter."

Buffer

A buffer is a memory sector that is used as a temporary storage location during input and output operations. The "preview buffer" length is programmable in DC8/DC FORENSICS, and is found in the Preferences section of the Edit Menu.

Butterworth Filter

A Butterworth Filter produces a maximally flat amplitude characteristic in the pass band or the reject band (depending on whether it is used as a Band-pass or a notch filter). It has a critically dampened response at the corner frequencies, having no ripple, and therefore it introduces little distortion into the signal that is feeding it. The Butterworth poles of signal transmittance are uniformly spaced on a semicircle, having its center on the imaginary axis. Its half-power frequencies are those at which the circle intersects the imaginary axis. The Low-pass, High-pass, and Band-pass filters offer the Butterworth response as an option.

Buzz

Buzz usually refers to a series of harmonics related to the frequency of the AC power mains. It differs from "Hum" in sound, because it usually contains a large number of higher frequency harmonics (created by phase controlled lighting dimmers or other non-linear systems connected to the power line). Buzz is best attenuated using the Harmonic Reject filter, the Spectral Filter or either the Expert Impulse or Narrow Crackle Filter(s).

Byte

An eight-bit word. Each sample of a monophonic .wav file is generally represented by two eight-bit bytes. Two eight-bit bytes are used to represent all of the integer numbers between 0 to 65,535 (actually +/- 32,768), which is the total dynamic range of DC8/DC FORENSICS when it is operating in 16-bit mode.
(1 Kilobyte (or 1 Kbyte) = 1,024 bytes)

Capacitance

(C)

Capacitance is the ratio of the electric charge given to a body compared to the resultant change of potential. It is usually expressed in coulombs of charge per Volt of potential change and its basic unit is the Farad. Energy is only stored (but not dissipated) in theoretical capacitance. Time

constants for audio filters are created with a combination of resistors and capacitors in various configurations. High-pass, Low-pass, Band-pass, and Notch filters can all be created with the appropriate combinations of resistors, capacitors, and operational amplifiers. The corner frequency for a simple first order RC filter = $1 / 2 \pi * (R \times C)$. The principle of capacitance (and conservation of charge) is involved in the operation of condenser and electret microphones and electrostatic loudspeakers and headphones.

$F = (\text{Farad})$ and $\mu F = \text{micro Farad}$

$\mu F = 1 \times 10^{-6} \text{ Farads}$

$pF = 1 \times 10^{-12} \text{ Farads}$

*Note: $\pi = 3.141592654$ (approximately)

Cassette Tape Equalization Time Constants

Compact Cassette tapes (which operate at 1 7/8 ips) commonly utilize one of the following two equalization time constants based on the tape type:

1. Normal (IEC Type 1) (Usually Ferrous Oxide based):
120 μSec .
2. High (IEC Type 2) (Usually Chromium Oxide based):
70 μSec

Cent

1,200 cents = 12 semitones = 1 Octave in the scale of "Just Intonation". In other words, the interval between two tones whose basic frequency ratio is the twelve-hundredth root of two is the "cent".

Cepstrum (Power)

Power Cepstrum displays are a way of graphically representing speech patterns using a specialized transform of an audio signal. It is not a reversible transform. It produces a unique view of the information relating various frequency bands associated with a signal and is useful for separating vocal tract information from the exciting pitch in voiced speech. Mathematically, the Power Cepstrum of a Signal = $|F\{\log(F\{f(t)\})\}|^2$. You can create cepstrum graphs using your DC Forensics version of this software package. The feature is found

under the Forensics Menu under the Voice ID feature.

Cepstrum (Complex)

A Complex Cepstrum is a reversible transform and can be useful for the graphic display of the vowel sounds produced by the human voice. It is also useful in identifying echoes created by acoustics in a room. Mathematically, the Complex Cepstrum = $FT(\log|FT(f(t))| + j2\pi m)$ where FT is the signals Fourier Transform and m is the integer value needed to unwrap the imaginary component of the complex log function. The horizontal axis is called the queffreny which is related to the rate of occurrence of a particular harmonic contained within the signal. This transform is implemented in the Voice ID feature found under the Forensics Menu of the DC Forensics Audio Laboratory version of this software.

Charge (Coulombs)

Charge in units of Coulombs is the product of Voltage time Capacitance ($Q = CV$). Also, refer to "Ampere" for the time relationship of the Coulomb..

Compact Discs

Compact discs are digital storage devices, typically 120 mm in diameter by 1.2 mm thick that are made of a polycarbonate compound. They can be used to store data, audio, video, and photographs. The active region of a compact disc begins at 46 mm from its center and runs to 117 mm. Reading of such discs is accomplished typically with an AlGaAs laser diode (producing light in the infrared region of 780 nm) in conjunction with a phototransistor sensor. There are a number of standards for data contained on Compact Discs. For a chart that describes these details, simply turn to our *Charts, Graphs and Other Useful info* pages.

Chebyshev Filter

A Chebyshev responding filter is one in which the pass-band produces a variation in attenuation between zero and its maximum value, and the band-stop requirement is only to increase monotonically to infinity. These filters display relatively steep roll-off characteristics, but, unlike the Butterworth filters, have ripple in their pass-band. Our Low-pass, High-pass, and Band-pass filters offer the Chebyshev response as an option.

Chirp Z-Transform

A Chirp Z-Transform (CZT) is used in the Diamond Cut High Precision Spectrum Analyzer to produce a higher resolution and signal processing speed compared to the Standard Precision Spectrum Analyzer. It is essentially a more advanced FFT technique or algorithm used to numerically calculate the Z-Transform of a sequence of a defined and limited number of samples relying on the fact that the values of the Z-Transform lie on a circular or spiral contour.

Classification of Amplifiers

Audio Amplifiers can be broken down into several classifications based on their degree of conduction relative to its input signal. The VVA Virtual Valve Amplifier utilizes two of the following classifications. The others are included in the description for completeness*:

Class A: The device or devices conduct for a full 360 degrees of the input signal. These amplifiers can be wired either in single-ended or push-pull configurations. Class A Audio amplifiers are usually used in pre-amplifier stages, or low power amplifier applications. This circuit has the poorest electrical efficiency, but produces predominantly even order distortion.

Class B: Two devices are operated out of phase with respect to one another. Each device conducts for only 180 degrees of the input signal. When the two amplified signals are combined, the full input waveform is represented, only amplified. This type of circuit is plagued by a phenomenon known as "crossover distortion" at low signal levels. This configuration is reserved for low performance PA amplifiers or AM (Amplitude Modulated) communications modulators. It is electrically efficient, but produces relatively large values of harmonic distortion especially at small signal levels.

Class AB: Two devices are operated out of phase with respect to one another, just the same as the Class B configuration. However, each device conducts for more than 180 degrees of input signal, but less than 360 degrees. This configuration produces a reasonable tradeoff between electrical efficiency and low distortion. This configuration is especially good at reducing crossover (notch) distortion (at zero crossing). It is commonly found used in

high power audio power amplifiers. Since the circuit is symmetrical, distortion levels can be quite low.

Class C: This configuration can consist of one or two devices which are conducting for anywhere between 90 to 180 degrees of the applied input signal. It is reserved for RF circuits only.

*Note: There are additional classifications of amplifiers involving tap switching, multiple rail, and pulse width modulation techniques, which have not been included in this listing.

Class D: Instead of operating power amplifier active devices in their linear range of operation, these same electronic devices can be used as state machines meaning that they are either on or off, and only in their linear region during transition (a parasitic state). Class D amplifiers operate in that manner, thereby minimizing power dissipation. The devices, which are acting as switches, are controlled by a pulse width modulator (PWM) and deliver power to the load via a Low-pass filter, which is used to attenuate the carrier or switching frequency. The rate of change of pulse width is proportional to the modulating audio frequency, while the magnitude of the change in pulse width is proportional to the modulating audio frequency amplitude.

Class G & H: These are "special case implementations" of classes A, B, and AB in which multiple power supply rails or variable rail voltages are used to minimize the power dissipation in the output devices while still allowing them to operate in their linear region.

Clipping

Clipping is a phenomenon, which occurs when a signal (or numerical value) exceeds a system's headroom. This concept applies to both analog and digital systems. The result of clipping is distortion. The amount of distortion produced depends on the amplitude of the over-driven signal. In DC8/DC FORENSICS, clipping will occur anytime a signal or calculation produces a numerical value greater than 2^{16} (or 65,536 counts or LSB's) when using it in 16 bit mode of operation. Clipping can be observed as a flattening of the slope (horizontal line) of a signal at its peak on the Source or Destination workspace displays.

CMRR (Common Mode Rejection Ratio)

CMRR, or Common Mode Rejection Ratio, is the ability of a balanced audio transmission system to reject in-phase signals that appear on the two input lines. It is usually expressed in dBu. For example, a CMRR figure of -60dBu means that 1 / 1000^{th} of a common mode signal presented to the transmission line is converted into a normal mode signal (which is the signal of interest).

Co-Axial Cable

A coaxial cable is one constructed in a manner in which the signal conductor is located in the center of the return conductor with a dielectric located in-between. This provides three notable characteristics for the cable:

1. The center conductor is shielded from the effects of "E" fields that may be present. "E" field coupled current is returned back to signal ground with little effect on the signal itself.
2. The loop area formed between the two conductors is very small compared to other types of conductors thereby minimizing inductance and also susceptibility to "H" field coupling.
3. The cable exhibits a characteristic impedance, which is independent of cable length (after past a few wavelengths) which is of a constant value related to its ratio of distributed inductance and capacitance. This makes the cable suitable for carrying RF (radio frequency) signals over long distances.

Co-Axial cables are often used to carry low-level signals from one audio device to another because of the first two mentioned characteristics.

Codec

Codec is an acronym for Coder-Decoder and pertains to the process of data compression and decompression. Audio examples include such algorithms as the Mp3, A-Law, Mu-Law, and the ADPCM formats. Video examples of Codec's are the various MPEG formats.

Columbia LP Equalization Curve

This equalization curve pre-dated the RIAA curve and was in use by the Columbia label and others in the early LP days of the 1950's. Here are the key frequency

inflection points for the early Columbia LP Curve:

1. 5310 uSec (30 Hz) (pull-out frequency)
2. 531 uSec (300 Hz) (turnover frequency)
3. 99.4 μ S (1600 Hz) (roll-off frequency)

This curve can be decoded with the Diamond Cut Virtual Pre Amplifier.

Comb Filter

A comb filter (or Harmonic Reject filter) is a wave reject filter whose frequency rejection spectrum consists of a number of equi-spaced elements resembling the tines of a comb. This filter is useful for getting rid of "Buzz" type noise containing more than just the line frequency fundamental component. In DC8/DC FORENSICS, it is called the Harmonic Reject Filter, and for more details, please refer to the same. The Spectral Filter found in the Forensics version of the software can also be used to create comb and inverse comb filters.

Compressor

A compressor is an electronic device that is used to reduce the dynamic range of an audio signal. They are often used to prevent overloading on certain mixer inputs (i.e. drums and vocals) in live performance applications. Radio stations often use compressors to make them "sound louder" when a potential listener is tuning across the radio band. This technique avoids violation of any FCC regulations regarding maximum % modulation or modulation index, yet still raises the perceived loudness of the station.

Corner Frequency

The corner frequency of a filter is the frequency at which the signal has been attenuated by 3 dB relative to the pass band region of the filter.

Crackle

Crackle is a term used to describe relatively low levels of impulse noise found on old phonograph recordings. It is very similar to impulse noise, only the peak amplitude is much smaller in comparison. Crackle sort of sounds like Rice Krispies just after you pour the milk in the dish. Crackle is usually caused by slight imperfections in the record-playing surface due to the use of coarse grain fillers in the record composition.

Sometimes, crackle is caused by gas bubbles that occur in the surface as the record "cured" after the stamping process. Crackle can be filtered out most effectively with the Impulse, Median, or Continuous Noise Filter. Very old acoustic recordings may be even more effectively de-Crackled (and de-Hissed at the same time) with the Average Filter.

Crest Factor

Crest Factor is the ratio of the peak value to RMS value of a signal (acoustical or electrical) over a defined time interval. For a 50 % duty cycle steady state square wave, this value is 1.0000. For a non-distorted sine wave, this value is $2^{\wedge}2$ (or about 1.4142). For audio signals, crest factor varies greatly depending on the material and the integration interval over which it is calculated. Classical music, for example, generally exhibits much higher values of crest factor compared to contemporary pop music when measured over an entire performance.

Crosstalk

Crosstalk is a figure of merit describing the degree to which one audio channel "bleeds" into another and expressed in dB. For example, a crosstalk figure of -60 dB indicates that the specified channel is affected by adjacent channels with an insertion ration of 1000:1.

dB (decibel)

1/10 of a Bel. A Bel is the basic unit for the measurement of sound intensity. It is a log scale measurement system used for relating the ratio of two acoustical or electrical parameters. Since electrical Voltage, current, and power are used to represent sound through audio signals, the following mathematical relationships may be found to be useful when relating them in terms of outputs and inputs:

dB (Voltage) = $20 \log V_{\text{output}} / V_{\text{input}}$

dB (current) = $20 \log I_{\text{output}} / I_{\text{input}}$

dB (power) = $10 \log P_{\text{output}} / P_{\text{input}}$

Note: A doubling of a Voltage or current represents a 6 dB change. A doubling of power represents a 3 dB change. For a table detailing the relationship between Voltage, Current and Power ratios in Decibels, turn to our Charts, Graphs and Other Useful Info section.

dBm

dBm is the power level of a signal expressed in dB, and referenced to 1 milliWatt (0.001 Watt).

Since:

$$V = (P \times Z)^{1/2}$$

wherein:

V = Voltage in Volts

P = Power in Watts

and

Z = Impedance in Ohms

therefore:

In a 500 Ohm audio line distribution system,

V at 0 dBm = 0.707 Volts

dBv

dBv is the Voltage level of a signal expressed in dB, and referenced to 1 Volt peak to peak. If a pure sine wave is the reference signal, its value would be 0.35 Volts RMS.

D-A Converter

A D-A Converter is a device to convert digital signals back into analog signals so that they will be compatible with analog sound reproduction equipment. DC8/DC FORENSICS requires a D-A converter with 16-bit resolution ($2^{16} = 65,536$). It also supports sound cards capable of up to 24-bit resolution.

DC-Art

Diamond Cut Audio Restoration Tools. Copyright 1994 - 2011, Richard A. Carlson and Craig P. Maier, Diamond Cut Productions, Inc. All rights reserved.

Diamond Cut Productions, Inc.

P.O. Box 305

Hibernia, N.J.

07842-0305

Email: www.diamondcut.com

DC Offset

A DC offset is a fixed value of Voltage that may have been added to a signal inadvertently. It contains no audio information. It can be eliminated by feeding

the signal through the DC8/DC FORENSICS high-pass filter set to 10 Hz and a slope of 6 dB/Octave. The High Pass Filter includes a Preset to eliminate DC Offset from a file.

De-Emphasis

The reversal of a pre-emphasis process. See Pre-Emphasis for more information.

De-Ess

De-essing is the process of decreasing the sibilance of over-modulated "ess" sounds produced by the human voice. The DC8/DC FORENSICS dynamics processor contains an algorithm for De-essing a signal containing this particular anomaly.

Diamond Cut Productions, Inc.

P.O. Box 305

Hibernia, NJ

07842

973-316-9111

www.diamondcut.com

Diamond Discs

The trade name for the records in the disc format produced by the Edison Company was "Diamond Disc." These records were cut vertically (hill and dale) and could only be played on Edison Diamond Disc phonographs designed for this purpose. They rotate at a speed of 80 RPM. To extract the vertical component of a signal provided by a stereo cartridge when transferring Diamond Discs, use the DC8/DC FORENSICS Mono (L-R) File Conversion feature.

De-Esser

A De-Esser is a non-linear system designed to attenuate the overly sibilant pronunciation of the letter "s" which can occur on some recordings. A De-Esser can also be used to reduce other forms of harmonic distortion which may be present on a recording. The Diamond Cut De-Esser is found within the Dynamics Processor routine and is enabled via a checkbox.

Differentiator

A differentiator is a device, system, or process which evaluates the rate of change of one parameter with respect to another. In its generic form, it is expressed as $f'(x) = dy/dx$. The reversal of the process of differentiation is Integration. The

Differentiator is found as one of the presets under the High Pass Filter.

DIM (Dynamic Inter-modulation Distortion)
Refer to TIM

Distortion

Distortion is a general term used to describe the undesirable effects that an audio system or process can have on the source input signal. There are many types of distortion, with some of them listed below:

1. Harmonic Distortion
2. Inter-modulation Distortion
3. Clipping Distortion
4. Crossover (Notch) Distortion
5. Phase / Jitter Distortion
6. Transient Inter-modulation Distortion (TIM) which is sometimes referred to as Dynamic Inter-modulation Distortion (DIM) or Slewing Induced Inter-modulation Distortion (SID).
7. Amplitude – Frequency distortion

Dither

In control loops, dither is the addition of a useful oscillation or noise signal into the system to overcome friction or hysteresis. This improves the response of the control loop to very small changes of the system reference signal. This principle has been extended to digital audio. In this case it implies the addition of a random noise signal inter-modulated with the LSB of the audio signal, effectively increasing the resolution of the system. Dither is included in the File Sample Rate Conversion feature.

Double-Ended Noise Reduction

Double-ended noise reduction involves encoding an audio signal when recording the same by some form of compression and reversing that process on playback. There are many different schemes for performing this function, and you can even create your own using the various algorithms contained within DC8/DC FORENSICS. The Dynamics processor or the Dynamic Noise Filter can be used to create various encoding / and or decoding schemes. It is important to match corner frequencies, thresholds, attack and release times in order to achieve good results. The rest is left up to the user.

Drive

Drive refers to the amplitude of a signal that is applied to an amplification device such as an electron tube or transistor. It represents the ac component, rather than the dc (or quiescent) component applied to the input. Since the above-described devices are intrinsically non-linear with regard to their transfer function, the larger the value of drive applied to the device, the greater will be the harmonic by-products. The DC8/DC FORENSICS Virtual Valve Amplifier allows you to adjust the drive level to the various amplifiers to vary the degree of "tube warmth". The program automatically compensates the output level, so that large values of drive do not produce substantial changes in overall system gain.

Dry

"Dry" is the term used to describe the signal output of a special effects generator (such as the Reverb) which contains only the non-processed signal. "Wet," on the other hand, refers to the effect signal alone. Like most special effect generators, the Reverb has an output mix control which allows you to transfer a signal from the effects generator which ranges from completely dry, to completely wet (no source signal), or to some mixture in between.

DTMF (Dual Tone Multi Frequency)

(Also known as Touch Tone)

DTMF is the dual tone encoding system used on the telephone system for dialing. Two frequencies are allocated for each number on a telephone keypad. For a chart that contains the Touch Tone dual frequencies, simply turn to the ***Charts, Graphs and Other Useful Info*** section.

Dynamic Filter

A filter in which its corner frequency is varied as a function of another parameter associated with the signal content of a sound source. Most often the corner frequency is that of a Low-pass filter that is modulated by the rectified output of a High-pass filter, although other schemes are possible. This sort of system changes bandwidth on the fly, and in co-ordination with the occurrence of high frequency content present in the source. It can be done either in a feedback or in a feed forward manner, with advantages and disadvantages attendant to each technique.

Dynamics Processor

An electronic device used to modify the characteristic dynamic amplitude response of an audio signal. These circuits can compress, expand, and De-Ess (remove overly sibilant "esses").

Dynamic Range

The dynamic range of an audio signal theoretically is the ratio of its smallest to its largest resolvable level or value. For a digital system, each bit represents a doubling of the signal so its dynamic range is simply its number of bits x 6 dB. In practice, with very high-resolution digital systems, other parameters may come into play such as the value of the systems noise floor, which may render some of the systems dynamic range unusable. For a table of common values of audio system resolution and their associated dynamic range, simple turn to our ***Charts, Graphs, and Other Useful Info*** section.

Ear

Your "ear" is the most critical piece of equipment that you will be using in the audio restoration process. It is important to realize that audio restoration is half science and half art. If you are only restoring audio for yourself, the audio restoration process is much less demanding as compared to situations where you may be performing the job for broad-based public consumption. In the second case, there are two very critical aspects of your "ear" which must be considered:

A. You must have a good sense of hearing. If you have a hearing deficiency, you may have a difficult time making the subjective judgments that are critical to the production of a commercially viable product, which will be acceptable to the "ear" of the general public. For example, if the "top-end" of your hearing is missing, it is more likely that you will produce restorations that seem harsh, hissy, and containing too many digital artifacts as far as the general public is concerned.

B. Even if you have an exceptional sense of hearing, you will need to develop a good "ear" for what the general public expects in terms of audio restoration. This requires good judgment, and a great deal of experience.

Edison, Thomas Alva

Thomas A. Edison was the inventor of the phonograph in 1877 at his laboratory in Menlo Park, New Jersey. Edison also invented the Carbon and the Condenser Microphone, and the "Edison Effect" which is the principle behind the early rectification and amplification devices that were used to develop the field of modern electronics.

Electrical Recording

"Electrical Recording" is the term given to a process which was commercialized around 1925 for mastering records in which microphones and electrical signal amplification was utilized to supply the energy required to modulate the cutting head stylus of the recording lathe. Prior to the invention of electrical recording, the acoustic energy of the various sound sources in the recording studio was the only source of energy that modulated the cutting head stylus. Electrical recording allowed more of the subtlety and detail of music to be captured on the wax master.

Electron Tube

The predecessor to the modern transistor was the Electron Tube (also sometimes referred to as an electron "Valve"). Dr. Lee DeForest invented the device around 1906 and called it the "Audion". He took a Fleming diode (a derivative of the Edison Effect light bulb - 1883), and installed a grid between its cathode and anode. He observed that small Voltage signals applied to the grid with respect to the cathode produced large current changes in the devices plate current. This device became known as the "triode", having three active elements within it. Thus was born the key device that became the foundation building block for the development of modern electronics, as we know it today. Electron tubes are basically amplification devices, which can be used in a myriad of applications. The Virtual Valve Amplifier uses the measured characteristics of real electron tube triodes and pentodes in various amplifier and rectifier circuit models to produce a versatile array of "tube-warmth" effects.

Elliptical Stylus

The shape of the tip of certain phonograph record playing styli which improves the high frequency response as compared to standard conical styli.

Engineering

The art of managing engines. (*Merriam-Webster*)

Envelope

The average amplitude of a .wav file as displayed in the DC8/DC FORENSICS workspace when zoomed-out.

Expander

An expander is a device that performs the opposite function of a Compressor. These devices increase the dynamic range of an audio signal source. When the process of compression is used in the recording or transmission process, and the process of expansion is used in the playback or reception process, the technique is known as companding. It is sometimes employed because it increases the signal-to-noise ratio of the analog recording or transmission process.

FFRR (Full Frequency Range Recording)

This is the equalization curve used on records marketed by London (Decca) Records. It claimed a frequency response of 50 to 14,000 Hz as early as the mid 1930's. Details can be found under "Equalization Curves."

FIFO

First In First Out or FIFO is a term used to describe the data flow in one form of a digital buffer. The first data into a FIFO buffer is the first data to exit out the other end of the buffer.

FIR (Finite Impulse Response)

FIR is a digital, non-recursive method for creating filters which can produce a phase linear response characteristic. FIR filters are always stable and are used in the Forensics Brick-Wall filter suite.

Fletcher-Munson Loudness Contours

The Fletcher-Munson Contours are graphs developed in the 1930's that identify the human perception of equal loudness as a function of frequency. Simply stated, the human ear provides the most perceptible flat response at very high loudness levels. At low loudness levels, the response falls off dramatically at the low end of the audio spectrum, and to a lesser degree, at its upper end. The curves indicate the region between 1,000 to 5,000 Hz are the flattest and most

independent of loudness. The flattest region lies between around 600 to 1,500 Hz.

FLAC (Free Lossless Audio Codec)

A file format that uses linear predictive coding techniques and provides lossless file compression. It reduces an audio file size roughly by 50%. FLAC is supported by Diamond Cut and is found under the File\Save As feature. Its file extension is .flac.

Flutter

A relatively rapid frequency modulation of the information on a recording due to rapid changes in the velocity of the record, tape, or the soundtrack of the source. Flutter is the rapid counterpart to Wow, occurring at a deviation rate in the range of 6 to 250 Hz. This distortion could have been introduced in the mastering process, or the playback process, or a combination of both. DC8/DC FORENSICS is not capable of correcting this sort of problem in a sound recording at this time.

Fractional Speed Mastering

Fractional speed mastering is the process of transferring a record at a slower speed to a new media, and then converting it to the proper speed at a later time. This has two potential benefits:

1. It allows persons who do not own 78 or 80-RPM turntables to make transfers of those types of records using their 45-RPM speed
2. It allows severely warped records to be transferred without skipping.

DC8/DC FORENSICS supports fractional speed mastering using the speed change filter. Presets will be found which accommodate various fractional speed-mastering situations.

Frequency Response

Frequency Response is the range of frequencies that a system will pass through without attenuation. The frequency response of audio equipment is generally specified with the upper and lower corner frequencies defined at the -3dB points. For most high performance audio system electronics, the frequency response will be at least as good as 20 Hz to 20 kHz +/-3dB.

However, loudspeaker systems rarely are able to reproduce the same spectrum within the specified linearity band.

Formants

Spectral concentrations of acoustical energy found in human vocal patterns produce resonances referred to as formants which can be used in the field of voice identification. Formants having the lowest frequency are designated as F0 and the sequence proceeds in an ascending rank up to Fn (usually between F1 to F4 in total number aside from the fundamental, F0). Comparing the relative amplitudes of these formants between a known reference and unknown samples can help establish the probability of a voice print match. Your Diamond Cut Forensics Audio Laboratory Spectrograph in conjunction with the Voice ID function (Find Formants) can be useful for plotting formant trajectories. These data are exportable for use in other software programs (like Excel) which may be useful for statistical analysis purposes.

Fourier Transform

A Fourier transform is a set of mathematical relationships which allow complex waveforms to be resolved into a series of fundamental frequencies, plus a finite number of terms which describe the waveforms harmonics. Fourier transforms are said to allow signals in the time domain to be represented in the frequency domain. Certain mathematical manipulations are more easily performed in the frequency domain as compared to the time domain, and DC8/DC FORENSICS takes advantage of this characteristic. After the mathematical manipulations have been completed, the results are re-converted back into the time domain via Inverse Fourier Transforms in order to re-create the processed version of the original time domain waveforms. This method is utilized in the Continuous Noise Filter.

Full-Duplex

In the context of audio restoration, the term "full-duplex" refers to a sound card that is capable of performing input and output functions simultaneously. For example, an analog sound card that has full duplex capability will be able to take an analog signal and convert it into a digital signal, at the same time that it is converting a separate digital signal into an analog form. The Live

Feed-through mode requires a full-duplex card in order for it to operate.

Gain

Gain is the amplification effect of an electronic system that is often expressed in decibels (dB). For example, an amplifier that has a Voltage gain of 20 dB produces an output Voltage signal that is 10 times greater in amplitude compared to its input. Many special effects audio processors produce "unity" gain. This implies that its output Voltage will be equal to the input Voltage (X 1 gain). Unity gain allows many signal processors to be placed in cascade without concern that the last processor in the chain will become overloaded due to the amplification build-up through each previous processor in the chain.

In general:

$$\text{Voltage Gain} = A_v = V_{\text{out}} / V_{\text{in}}$$

or -

$$\text{Voltage Gain in dB} = A_v (\text{dB}) = 20 \log V_{\text{out}} / V_{\text{in}}$$

Total System Gain in dB = Subsystem #1 Gain in dB + Subsystem #2 Gain in dB + Subsystem #N Gain in dB (when the subsystems are connected in a cascaded configuration).

Note: If the subsystem gains are not given in dB, the total system gain is the product of the various subsystem gain values. For example, the total Gain = (Subsystem #1 Gain) X (Subsystem #2 Gain) X (Subsystem #N Gain).

The gain (A) of an electrical system can be given in terms of any of the following:

1. Voltage: ($A_v = V_{\text{out}} / V_{\text{in}}$) --- (Voltage Gain in dB = $20 \log V_{\text{out}} / V_{\text{in}}$)
2. Current: ($A_i = I_{\text{out}} / I_{\text{in}}$) --- (Current Gain in dB = $20 \log I_{\text{out}} / I_{\text{in}}$)
3. Power: ($A_p = P_{\text{out}} / P_{\text{in}}$) --- (Power Gain in dB = $10 \log P_{\text{out}} / P_{\text{in}}$)

Generational Loss

Each time an audio signal is transferred from one medium to another, it will suffer some degree of "generational loss." These losses include noise buildup, distortion, phase jitter, quantization errors, etc. In

analog systems, generation loss is much more of a significant factor in signal degradation compared to that, which will be found in digital systems. In practical terms, analog signal transfers should be minimized in audio restoration work. The best results are produced if the analog to digital conversion process is performed only once. Ideally, the only analog process would be the original A-D transfer. Once in the digital domain, all processing, including the final transfer to DAT or CD-R, can be performed by your computer and the DC8/DC FORENSICS program. The only future conversion back to the analog world would occur during the playback process of the CD or the DAT through your audio system.

Graphic Equalizer

A Graphic Equalizer is a signal processor in which the audio band is divided into smaller spectral bands (portions). Each spectral band can be adjusted in terms of either the gain or the attenuation of the frequencies that fall within that band. Most Graphic Equalizers are Octave based, and contain about 10 bands. However, some are 1/3 Octave based and have about 30 bands. Octave based graphic equalizers (including the one contained within the DC8/DC FORENSICS application) typically break the audio spectrum down into bands with the following center frequency values:

31 Hz, 62 Hz, 125 Hz, 250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz, 8 kHz, 16 kHz

Your DC EIGHT software program also includes 20, and 30 Band Graphic Equalizers. The Forensics version includes a 32,000 band EQ called the Spectral Filter.

Ground Loop

A potentially detrimental loop formed when two or more points in an electronic system that are nominally at ground potential are connected by a conducting path. The term usually is employed when, by improper design or by accident, unwanted noisy signals are generated in the common return of relatively low-level (audio) signal circuits by the return currents or by magnetic fields generated by relatively high-powered circuits or components.

Harmonic Exciter

A Harmonic Exciter is an electronic device or algorithm, which synthesizes odd and/or even harmonics of the upper end of the audio spectrum presented to it, and then re-inserts them back into the signal path. This device will "liven-up" older recordings in which the upper musical registers are missing due to generational losses or lack of response to begin with. It can also be used to enhance vocals, or stringed instrument recordings. The Exciter is found under the Virtual Valve Amplifier (VVA) system located under the effects Menu. It uses real models of Electron Tube rectifier and amplifier circuits to accomplish its synthesis.

Harmonics

Harmonics are the odd and even multiples of a fundamental frequency. In music, it is the distribution of these harmonics that provides the characteristic (or timbre) that gives each musical instrument or human voice a unique sound.

Harmonic Distortion

Harmonic Distortion results from the interaction of a non-linear transfer function of a system on a signal. The non-linearity of the system creates undesirable harmonic products (except in rock and roll) that modify the sound of the original signal. Devices like transistors, vacuum tubes, microphones, phonograph cartridges, loudspeakers, and A to D converters all have non-linearities to some degree. In some cases, feedback is used to correct for non-linearity and in other cases using the device only in a very limited portion of its total dynamic range is the method for minimizing the production of harmonic distortion. For information on the measurement of this parameter, see THD (Total Harmonic Distortion).

DC8/DC FORENSICS can produce signal distortion when one of the algorithms attempts to drive the system to full scale or beyond. It is therefore necessary to be careful when applying the Gain Change or the Graphic Equalizer algorithms, both of which can increase the gain of the system causing signals to exceed the programs dynamic range. The distortion produced as a by-product of this mechanism is called clipping.

Harmonic Reject Filter

Please refer to "Comb" or "Multiple Notch Filter."

Helmholtz Resonator

A Helmholtz resonator is the most basic resonant system found commonly in musical instruments and consists of an enclosed volume of air and an opening (or aperture). It is the acoustical analogue to the electrical LC resonant tank circuit.

Hertz(Hz)

Hertz is a unit for the measurement of frequency. 1 Hertz = 1 cycle per second.

Heterodyne

When two frequencies mix with one another through a non-linear system, sum and difference signals are produced. These signals are referred to as heterodynes and are also sometimes referred to as "beat frequencies." Early audio oscillators constructed before 1935 used the heterodyne technique for producing output signals in the audio range by "beating" two RF (radio frequency) oscillators against one another. They were referred to as Beat Frequency Oscillators (BFOs,) and were eventually replaced with a gain-stabilized form of the Wien Bridge oscillator. Spurious heterodyne signals are produced on the various AM radio bands due to adjacent channel interference. In the US, these signal are 10 kHz, and in Europe, they are 9 kHz. The DC8/DC FORENSICS notch filter can be used to remove them.

Hill and Dale

See Vertical Cut

High-pass Filter

A filter that attenuates all frequencies that fall below its corner frequency. The degree of attenuation of a signal outside of the filters pass band depends on the frequency of interest, and the corner frequency and slope (order) of the High-pass-filter. This type of filter is often used to reduce rumble and muddy bass on a recording.

Hiss

Random noise at the top end of the audio frequency spectrum is often referred to as hiss. Generally this is considered to be the random noise that is heard above 5 kHz. A good example of "hiss" is the sound you will hear if you tune a FM tuner to the top or

bottom end of the band where there are no stations transmitting with the "mute" button disabled. (This is a form of limited bandwidth "white noise.")

Hum

Noise introduced into a recording or sound system that is harmonically related to the power line frequency. In the US, this will be 60 Hz and in Europe, this will be 50 Hz., and in both cases, it will include harmonics of the line frequency. The most common "hum" frequencies are the fundamental (usually due to ground loops) and /or its second harmonic (due to defective power supply filter capacitors in electronic equipment). To attenuate Hum on a recording, use the DC8/DC FORENSICS Notch filter set to either 50 or 60 Hz, depending on the hum frequency. Start with a bandwidth setting of around 0.2 Octave. Adjust the bandwidth to the minimum value required to effectively attenuate the Hum. This will minimize the Notch filters effect on all other frequencies.

Human Voice Frequency Range

See "Voice Frequency Range"

IIR

IIR is the acronym for "Infinite Impulse Response" which is a form of recursive filter. It employs, in the simplest example, a simulation of at least one resistor and one capacitor in a circuit, which are represented by a pair of recursive equations. The types of filters employing IIR techniques in the "Filter" menu are such functions as the Low-pass, High-pass, Band-pass, notch, and slot filters as well as the graphic and parabolic equalizers.

Impedance

(Z)

The total opposition including resistance and reactance which a circuit element(s) offers to the flow of an alternating current, measured in Ohms.

$$Z = ((R^2) + (Xc^2) + (XI^2)) ^ {1/2}$$

Wherein:

Z = Impedance in Ohms

R = Resistance in Ohms

Xc = Capacitive Reactance in Ohms

XI = Inductive Reactance in Ohms

Some standard Input and Output Impedance values that you will encounter are as follows:

1. 1 Ohm - The basic unit of measurement for Electrical Resistance, Impedance, or Reactance
2. 2 Ohms (sound re-enforcement systems), 3.2 Ohms (antique audio), 4, 8, 16, and 32, Ohms, (standard Loudspeakers)
3. 8 Ohms – The most common loudspeaker impedance found in the US in 2002.
4. 50 Ohms - Standard Unbalanced Co-Axial impedance for RF signal transmission
5. 75 Ohms - Standard Unbalanced Co-Axial impedance for Television and FM signal transmission
6. 300 Ohms - Standard Balanced impedance for Television and FM signal transmission
7. 377 Ohms - Impedance of Free Space
8. 500 Ohms - Standard Balanced Microphone impedance
9. 600 Ohms - Standard Telephone Exchange Audio line impedance
10. 2,000 Ohms - Antique Audio (headphones & 1920's vintage horn loudspeakers)
11. 20,000 Ohms - Common single ended input impedance found on Professional Audio Equipment
12. 47,000 Ohms - Common Magnetic Phono Cartridge Loading Impedance
13. 50,000 Ohms - Standard Unbalanced High Impedance Microphone Impedance.
14. 100,000 Ohms - Common Input Impedance on Audio Equipment
15. 1 Meg Ohms - De-Facto Standard, Oscilloscope Input Impedance
16. 10 Meg Ohms - De-Facto Standard, True RMS Voltmeter Input Impedance

Impulse

Mathematically, an impulse function is an event of infinite amplitude, and infinitesimal time duration. In EIGHT/DC FORENSICS terms, an impulse is a transient that begins and ends within somewhere between 50 μ S to 1 mS, with amplitudes which are generally higher than the average program material in a .wav file.

Inductance

(L)

The inductance of a circuit component (most often a coil) is the rate of increase in magnetic linkage with an increase of current. The unit of measurement of inductance is the Henry which corresponds to a rate of linkage increase of 10^8 Maxwell-turns or one Weber-turn per Ampere of current. Energy is stored (but not dissipated) in theoretically ideal inductors. The principle of inductance is a strong element in the operation of electronic transducers such as loudspeakers, magnetic phono cartridges, dynamic microphones, and transformers. Resonant circuits can be created utilizing a combination of capacitors and inductors. The basic resonant frequency of such a circuit is given by $F_r = 1 / 2 \pi (L \times C)^{1/2}$. This principle can be used to create narrow Band-pass and notch filters.

The unit of measurement of inductance = H (Henry)

Note: $\pi = 3.141592654$ (approximately)

.ini files

This is where the initialization constants, factory presets, and user presets were stored by the software in all earlier versions of the software. Because of the 64 Kbytes total .ini file length limitation; a special presets directory is presently used for this function allowing a much larger number of total presets. The old .ini file extension used to be found in the Windows directory. EIGHT/DC FORENSICS uses the system registry for all other settings storage.

Instantaneous Sound Recordings

Instantaneous sound recordings are ones in which playback can immediately follow the recording process. The first sound recording made by Edison was "Instantaneous." The following is a brief history of this form of recording technology:

- Wax Cylinders (1877 forward)

- Aluminum Discs (mid 1920s)
- Pre-Grooved Plastic Discs (early 1930s)
- Acetate covered Aluminum Discs (mid 1930s)
- Acetate covered Glass Discs (early 1940s)
- Magnetic Wire (mid 1940s)
- Magnetic Analog Tape (late 1940s forward)
- Digital Audio Tape (mid 1970s forward)
- CD/R's (early 1990s forward)

Integrator (\int)

An Integrator is a device, system or process which evaluates the area under a curve of one function vs. another. It is the inverse process of Differentiation (also refer to "Differentiator"). The Integrator is found as one of the presets under the Low Pass Filter.

Inter-modulation (Distortion)

Inter-modulation distortion is a very specific type of distortion that results from the amplitude modulating effect which one frequency has on another in a non-linear system. This form of amplitude modulation results when one frequency appears as the "carrier" and the other as the "modulating" signal. Since IM distortion is due to non-linearity's in the transfer function of a system creating an AM effect, this results in the production of frequency components that are equal to the sums and differences of integral multiples of the components of the original complex wave. A two-tone test source is used to measure this parameter, typically consisting of (approximately) a 60 Hz test signal added to a 7,000 Hz test signal feeding the device or system under test. The low frequency signal is applied to the system being tested with an amplitude value much higher than the upper test frequency, typically by a factor of four or more to one. Using 60 and 7000 Hz test signals as an example, vestiges are measured at 6,040 and 7,060 Hz and divided by the value of the 7,000 Hz fundamental in order to determine the level of Inter-modulation Distortion.

Inertia

The property by which matter that is at rest will tend to remain at rest, and matter that is in motion will tend to remain in motion (in the absence of friction).

I/O

Input / Output refers to the ports into which electronic signals are fed to an electronic device and the ports from which electronic signals are derived from an electronic device. EIGHT/DC FORENSICS allows you to choose between several I/O ports, provided you have the sound cards to support the feature.

IPS (Inches per Second)

The linear velocity of magnetic tape moving past a recording or playback head is referred to in terms of its IPS (inches per second) value. For a table of common tape deck speeds, please refer to our *Charts, Graphs and Other Useful Info* page.

kHz (kilo Hertz)

The unit used in the measurement of frequency equal to 1000 Hertz. In earlier times, this term was Kilocycles (per second).

kOhms

The unit used in the measurement of electrical resistance equal to 1000 Ohms.

Latency

Latency is the delay time encountered when operating in Preview mode or in "Live" feed-through mode. Maximizing the speed of your computer system minimizes latency.

Lateral Cut

A record recording technique in which the groove modulation (undulations) occurs in a side-to-side direction, as opposed to up and down. This technique was popularized by Emile Berliner with his "Victrola" phonograph.

Launch

Starting a program. This is accomplished by double clicking on the appropriate Icon.

Least Significant Bit (LSB)

The smallest quantified increment which an Analog to Digital or Digital to Analog Converter can resolve an analog Voltage or current. (LSB's are sometimes referred to as "counts.") In EIGHT/DC FORENSICS, this value is 1 part in 65,536 (or 1 part in + / - 32,768), for 16 bit system applications. For 24 bit system applications, this value increases to 1 part in 16,777,216 (or 1 part in + / - 8,388,608). For other sampling depths (values of resolution) use the formulae:

$$+/- \text{LSB's} = (2^{\wedge}(\text{SD})) / 2$$

wherein SD = Sample Depth (or resolution) in bits

Limiter

An electronic circuit or system consisting of non-linear elements that will not allow signals above a threshold value to pass through to its output. An upward compressor will produce this effect when its ratio is set to a high value. The Dynamics Processor can be used as a signal limiter when used in compressor mode when high values of "ratio" are selected.

Lissajous Figures

When two sine waves are displayed on an X-Y display, with one applied to the X-axis and the other to the Y-axis, the interacting vectors of the two waveforms are displayed. These waveforms are referred to as Lissajous figures. Signals having the same frequency but of differing phase (other than 180 degrees) will form elliptical patterns, the phase of which can be calculated from the intercepts of the waveform with the display axis. This technique if often used to adjust the azimuth of tape recorder recording and playback heads. A properly aligned tape head will produce no ellipse, but only a 45-degree line with a positive slope. This can be done using the Time Offset feature found in the File Conversions Filter in conjunction with the X-Y plotter found under the View menu.

Loudness

The loudness of a sound is the perceived magnitude due to the auditory sensation produced by an acoustic signal. It is a function of frequency and signal amplitude. A 50 Hz tone requires sound field intensity 250,000 times larger than a tone at a reference of 1,000 Hz to achieve the minimal level of perception. The unit of measurement for loudness is the Sone. See Sone for more details.

Low-pass Filter

A filter that attenuates all frequencies that fall above its corner frequency. The degree of attenuation of a signal outside of the filters pass-band depends on the frequency of interest, the corner frequency, and slope (order) of the low-pass filter. This type of filter is often used to reduce the hiss on a

recording. However, low-pass filters will also attenuate the "highs" on a recording at the same time, which make them generally undesirable for this application.

Magnetic Phono Cartridge

A magnetic phono cartridge is a device for converting the mechanical motion of a record stylus into electrical signals utilizing the properties of magnetic circuits. There are three types of magnetic phono cartridges. They are:

1. Variable Reluctance (early magnetic cartridges)
2. Moving Magnet (the most commonly used)
3. Moving Coil (quite expensive and having costly stylus replacement)*

* The output impedance of moving coil (MC) cartridges are in the 10 to 100 Ohm range. Therefore, they require special matching transformers or pre-pre-amplifiers in order to be able to drive a conventional magnetic cartridge input on an audio pre-amplifier.

MByte

One million Bytes. (Sometimes 1024 kBytes for disks)

Median

The median value of a series of numbers is the number that resides in the center of the sorted string. For example, in the series of numbers 2, 4, 7, 3, 8, 0, 7, 1, 9, the median value is **4**. (sort = 0,1,2,3, **4**,7,7,8,9)

Milliseconds

The unit used in the measurement of time equal to 1/1000 of a second.

Mils

The unit used in the measurement of distance equal to 1/1000 of an inch. The diameter of phonograph styli are generally specified in "mils." (If the stylus is elliptical in shape, the larger of the two dimensions is generally given.)

Modulation

An electronic process in which one source modifies the characteristics of another signal source. For example, an audio signal may be used to Amplitude, Frequency, or Phase Modulate a sine wave signal (called a carrier). The result would be an Amplitude

Modulated carrier in the first case (AM). In the second case, the result would be a Frequency Modulated carrier (FM). In the last case, the result would be a Phase Modulation (PM) carrier. These are techniques used for transmitting radio, television, and data. Sometimes, in audio, one refers to the undulations on a record as record groove "modulation."

Monophonic

An audio signal or a .wav file that contains only one unique channel of aural information is sometimes referred to as being monophonic.

μ (μ)

The small signal amplification factor that a device exhibits in a circuit, often associated with electron tubes.

Voltage Amplification = $\mu \times R_L / (R_L + R_p)$

Where μ = Tube Amplification Factor

R_L = Plate Load Resistance in Ohms

R_p = Plate Resistance in Ohms

Mu-law (μ -law) Compression

This is a form of data compression primarily used in telephonic applications. It takes advantage of the logarithmic nature of the sense of human hearing to accomplish its task effectively. It compresses a 16-bit signal into 8 bits using logarithmic signal mapping, producing a 2:1 compression ratio while maintaining 13 bits of dynamic range. The bottom 3 LBS's of precision are dropped. Typically, this is used in 8 kHz Monophonic formats. DC EIGHT supports this format.

Multi-path Distortion

Multi-path distortion is a phenomenon that can occur during FM broadcast reception. It occurs when the receiving antennae pick up two signals from the same transmitter. This dual pickup consists of the direct signal from the transmitter (usually a line of sight trajectory) and a second parasitic signal arriving at the antenna some time later. The second signal is a reflected signal off of a mountain, building or other object, and arrives at the antennae some time after the main signal had arrived. The time shift between the main signal and the reflected signal creates phase distortion of the demodulated audio signal when these two signal mix together. This phase distortion manifests itself in the last two octaves of the

audio spectrum and sounds like "slurring" of the pronunciation of the letter "s" and general harshness. It will sound worse on a stereo broadcast than on a monophonic one. There are several cures for this problem. Purchase a directional antennae (one with a high front to back ratio) and install it as high as possible, aiming it towards the transmitter of interest. Secondly, you can minimize the problem by switching over to monophonic during a particularly distorted broadcast. And lastly, when all else fails, you can reduce the distortion by utilizing the "de-esser" found in the Dynamics Processor.

Multiple Notch Filter

The term used in the EIGHT/DC FORENSICS program used to describe a comb filter. A comb filter is a wave reject filter whose frequency rejection spectrum consists of a number of equidistant elements resembling the tines of a comb. This filter is useful for getting rid of "Hum" type noise containing more than just the line frequency fundamental component. This type of noise is line frequency related noise and sometimes described as "Buzz." This results from the interaction of non-linear systems with the finite output impedance presented by the power line sine wave Voltage waveform, adding harmonics to the same. Buzz can also be introduced into and audio system through non-sinusoidal current waveforms producing "H" fields which couple into noise sensitive loop areas (or ground loops) in audio systems.

Musical Scale

There are two relatively common musical scales. They are the "Scale of Just Intonation", and the "Scale of Equal Temperament". The Scale of Just Intonation requires at least 30 discrete frequencies for each octave, making it relatively impractical to build musical instruments with fixed tones to play in the Just Scale. Therefore, the scale of Equal Temperament containing only 12 notes per octave is the one in general use.

For a chart which displays the frequencies of four octaves of the tempered scale, simply turn to our *Charts, Graphs, and Other Useful Info* section.

NAB Equalization Curve

(National Association of Broadcasters)

The NAB Curve is a set of equalization frequency response contours, which are used by manufacturers of analog tape recorders to compensate for the inductive nature of a tape head. The equalization time constants specified depend on tape speed. One pair of time constants are specified for 1 7/8 ips (inches per second) and 3 3/4 ips. Another pair of time constants are specified for 7 1/2 ips and 15 ips. The low frequency breakpoint for all speeds is 50 Hz. The high frequency breakpoint for 1 7/8 and 3 3/4 ips is specified as 1770 Hz. The high frequency breakpoint for 7 1/2 and 15 ips is specified as 3180 Hz.

Neper (Napier) (Np)

Nepers are units of ratios of measurement like the dB, except using base e (~2.71828183) rather than base 10. For details, please refer to "dB" in this glossary. To convert between Nepers and dB, use the following relationships:

$$1 \text{ Np} = \text{dB} \times 0.115129255$$

or

$$1 \text{ dB} = \text{Np} \times 8.685889638$$

Noise

Unwanted disturbances superimposed upon a useful signal that tends to obscure its information content. Also, refer to Signal-to-Noise ratio for more information.

Noise Gate

A noise gate is an electronic device, which turns off a signal path when an input signal is below a predetermined threshold value. The Dynamics Processor produces a noise gate effect when you check the Expander/Gate function. You must set the ratio to the highest number for the best noise gate effect.

Notch Distortion (crossover distortion)

A discontinuity in a signal waveform sometimes produced by power amplifiers is referred to as "Notch Distortion". It is always found in class B Amplifiers and sometimes found in inadequately biased class AB Amplifiers. It is usually associated with audio power amplifiers. High levels of "Notch" distortion results in a raspy sound at low signal output levels.

Notch Filter

A notch filter is one which attenuates all frequencies close to the center frequency of the filter setting. The degree of attenuation and the range of frequencies which are attenuated by this filter are determined by the filters Q or bandwidth. This type of filter is often used to minimize hum or acoustic feedback from a recording. This type of filter is sometimes referred to as a "band reject filter."

Octave

An octave is a group of eight musical notes and also a doubling of frequency. For example, the range of frequencies from 440 Hz to 880 Hz is 1 octave. The next octave will end at 1760 Hz. Note that in two octaves, the frequency has increased by a factor of four.

Offset

A fixed or DC value of Voltage or current added into a circuit to shift the quiescent operating point of a device or display. Offset is used in EIGHT/DC FORENSICS to allow detail to be seen in a signal when the detail exists towards the top or bottom of the signal workspace display area. DC offset can be removed from a .wav file by applying the high pass filter set to 20 Hz, Butterworth response, and 6 dB/octave.

Ohm (Ω)

(R) (or the Greek Letter Omega Ω)

The Ohm is a unit of electrical resistance in which a potential difference in a circuit of 1 Volt produces a current flow of 1 Ampere.

Ohms Law

$$V = I \times R$$

wherein:

V = Voltage in Volts,

I = current in Amperes,

R = resistance (in Ohms)

Over-modulation

When an audio signal is applied to an audio device, which is greater than the device can handle in a linear transfer manner, this creates a condition of "over-modulation." It

results in a distorted sound in the output of the device being over modulated. Sometimes, this condition is referred to as "clipping," meaning that the amplification devices of an electronic system are either cutting-off or saturating due to overdrive.

Overtones

Overtones are multiples of a fundamental frequency. The Diamond Cut Overtone Synthesizer is capable of creating signals of this type at the upper end of the audio spectrum (above 6000 Hz). The Virtual Valve Amplifier (and its Exciter) are also capable of creating overtones referenced to various fundamental frequencies.

Parametric Equalizer

A variable electronic filter in which the following three parameters may be adjusted on each parametric channel:

1. Frequency
2. Level (attenuation or amplification)
3. Bandwidth

Parametric equalizers are usually equipped with several parametric channels, which can all be used simultaneously or each one can be individually bypassed.

EIGHT/DC FORENSICS includes a 10 band Paragraphic equalizer, which is a combination of a parametric and a graphic equalizer.

Pathé

Pathé Freres Phonograph Company was a European based record and phonograph company, who utilized a somewhat unique groove modulation technique. Their method produced a vertical stylus displacement (like Edison Hill and Dale Diamond Discs and Cylinders) however; this was accomplished by a different mechanism. The groove on these recordings is "width" modulated, and so when a conical stylus interacts with these groove width modulations, a vertical displacement is thereby produced. If you are transferring a Pathé 78 rpm recording with a stereophonic pickup cartridge, you will need to use the EIGHT/DC FORENSICS Mono (L - R) file conversion algorithm.

Pentode

A Pentode is an electron tube (or valve) containing five elements. They include a cathode, anode, control grid, screen grid or beam deflector electrode, and a suppressor

grid. They are most commonly used in audio power amplifiers, but are sometimes found in microphone pre-amplifiers. Typical beam power pentodes listed in ascending power levels include types 6BQ5/EL84, 6L6GC, 5881, 7591, KT-66, 6CA7/EL34, KT-88, and 6550.

Phantom Microphone Power

(Sometimes referred to as P48)

High performance microphones (such as large diaphragm condenser types) require an external source of power to drive their internal circuitry. The Voltage of this power source is generally between 12 to 48 VDC (most commonly 48 VDC). The negative side of this power source is connected to pin 1 (the shield side) of a 3 pin XLR style connector. The positive side of this source is connected (via two ~ 6800 Ω resistors) to pins 2 and 3 on the connector with one resistor located in series with each leg at the source end.

Phase Inversion

Phase inversion is the phenomena when one of two signals has become 180 degrees phase shifted with respect to the other. This sometimes accidentally occurred on vinyl stereo recordings because the input leads to one of the two cutting lathe driver heads became "swapped" in location. This can be corrected by using the File Converter, using the Left or Right Phase-Invert feature.

Phon

The Phon is a measurement of the perceived loudness of a sound which takes into account the non-flat frequency response of the human sense of hearing. The perceived loudness of a sound in Phons is equal to the sound intensity in dBs of an equally loud 1 kHz pure tone signal; 1 dB SPL @ 1 kHz = 1 Phon. Also, refer to the glossary topic "Fletcher-Munson Loudness Contours".

Pi(π)

Pi (Greek Letter) is the symbol that relates the ratio of the circumference to the diameter of a circle.

$$\pi = C / D$$

wherein:

C = Circumference of a Circle

D = Diameter of a Circle

$\pi =$

3.1415926535897932384626433832795029

(approximately because it is an irrational {non-repeating} number)

Pink Noise

Pink Noise is random noise, which is characterized as containing equal energy per unit octave. When viewed on an octave based spectrum analyzer, it will produce a flat horizontal line on the display. Pink Noise is useful for characterizing the frequency response of electronic systems and for analyzing room acoustic transmittance and resonance. Pink noise can be created through a two-step process using DC 8 or DC Forensics 8. First, create white (random) noise with the Makes Waves generator function. Next, process the signal through the Multifilter using the factory preset labeled "White to pink noise converter, 20 kHz." You can also convert Pink Noise to White Noise by applying the Multifilter preset called "Pink to White Noise Converter" to a Pink Noise File.

.pkf files

These are "peak files" (.pkf) which represent the .wav files for display purposes in the source and destination windows. They are a small subset of the .wav file consisting of one sample for every 200 .wav file samples. The data point selected is the peak value of the .wav file found within that 200 sample window. Each .wav file is stored with an attendant .pkf file that is created by the software at the time when the file is recorded or modified or when the "rebuild peak file" command is used.

Power

Power is the time rate for the transfer of energy in any system. In other words, Power = Energy / time. In electrical terms, power is given in Watts and has the following relationships to Voltage, Current, and Resistance:

$$P = V \times I (\cos \theta)$$

wherein:

P = Power in Watts

V = Voltage in Volts

I = Current in Amperes

θ = the displacement phase angle between the Voltage and Current Waveforms (assuming that they are both sine waves)

also,

$$P = (I^2) R$$

and

$$P = (E^2) / R$$

wherein:

R = Resistance in Ohms

Power Amplifier

(Power Amp)

A power amplifier is a device that provides power amplification of an audio signal. Generally, this is the device that is used to drive a loudspeaker, the cutting head of a record lathe, or an audio transmission line, and is the final stage of amplification in an audio system. Audio power amplifiers generally develop somewhere between 10 to 1000 Watts of output power, depending on make and model (although shake table audio amplifiers and AM radio transmitter modulators can be found which produce well over 50,000 Watts).

To minimize power loss in the transmission process, and to maximize the systems dampening factor, it is important to minimize Voltage drops across loudspeaker distribution cables. Poor dampening factor can produce an ill-defined bottom-end (bass). Long distances between your power amplifier and your speaker system will require larger diameter cables. To determine the correct cable for your application, refer to the Wire Table provided in this Glossary.

Power Cepstrum

See "Cepstrum"

Pre-Amplifier

(Pre-amp)

A device that provides Voltage amplification of an audio signal. Sometimes these devices also include equalization networks and/or tone (bass, treble, loudness, etc.) controls. Pre-Amplifiers generally

produce about 100 milliwatts of output power and require a power amplifier connected in cascade in order to be useful for driving loudspeakers.

Pre-Emphasis

The intentional added amplification which is sometimes applied to the top end of the audio spectrum during a recording or radio transmission process in order to raise the signal level at high frequencies substantially above the noise level of the system. This process is reversed during the reproduction process of the signal in order to recreate an overall flat frequency response. The result of this process is an improvement in the signal-to-noise ratio of the system. For example, the third specified time constant of 75 μ Sec associated with the RIAA equalization curve is pre-emphasis. Also, FM broadcast transmission utilizes a 75 μ Sec (or sometimes a 25 μ Sec) pre-emphasis to improve its signal-to-noise ratio. This process is reversed at your receiver (de-emphasis). The Paragraphic equalizer contains 75 μ Sec pre-emphasis and de-emphasis preset curves.

Presets

Most of the filters and effects have a plethora of descriptive presets. Most often, the most efficient place to start when using a particular filter or effect would involve selecting one of the factory presets, and then tweaking the parameters to fine tune the system to your own personal taste. If you desire to keep a separate copy of your presets on diskette, it can be found in the Diamond Cut directory under *.DefaultPresets.

Pure Tones (as related to Noise Laws)

A pure tone occurs when the noise level in any one Octave weighted frequency band exceeds those in an adjacent frequency band by 3 dB or more. The High Precision Analyzer found in the DC Forensics version has the characteristics needed to evaluate compliance with the various state and federal laws in this regard when used in conjunction with a calibrated / certified / NIST traceable microphone.

Q

(Quality Factor of Resonant Systems)

Q is the ratio of the reactance of a system (or filter) to its resistance (or losses). Q

determines the systems bandpass width. Higher Q values producing sharper (narrower and more selective) responses.

$Q = \text{Reactance (@ system resonance)} / \text{Resistance.}$

Quefrency

Quefrency is the unit of measure associated with the horizontal (X) axis of a complex cepstrum graph. The units are not frequency or time in the standard sense of those terms. Rather it is the ratio of the sample rate divided by the number of samples that a peak persists. For example, if a signal persists for 50 samples and is sampled at 48,000 samples per second system, its quefrency will be $48000 / 50 = 960$ Hz.

Quiescent Point

The Quiescent point (or operating point) of an amplification device like an electron tube or a transistor, refers to the bias established on its linear portion of the transfer function curve when the device is "at rest" (i.e. no signal input applied). The Virtual Valve Amplifier allows you to adjust the Quiescent (operating) point of class A amplifiers anywhere from near cutoff to near saturation.

RAM (Random Access Memory)

RAM is a digital electronic device for storing binary information temporarily. RAM performance is generally characterized in terms of its size in MBytes, and its access time in nanoseconds. Your computer will need a minimum of 8 MBytes of RAM to run the EIGHT/DC FORENSICS application correctly.

Real Time

A system that can process a signal and output the signal at the same rate at which it is being fed into the system is said to be a real-time processor. The EIGHT/DC FORENSICS algorithms can process signals in real-time or faster provided your platform is a 200 MHz Intel Pentium or higher. The exception to this rule is the 200 MHz Intel Pentium-Pro processor. Since it is not optimized for 16 bit applications, it cannot run all algorithms in real time or faster.

Real Time Analyzer (RTA)

A Real Time Analyzer is a form of spectrum analyzer used for the analysis of audio signals. Unlike conventional spectrum analyzers, it does not use a single filter in a scanning mode to produce an Amplitude vs. Frequency display, which is a relatively slow process. Instead, it processes audio signals in parallel, so that all frequency bands are displayed simultaneously. Generally, RTA's have 31 bands (in 1 / 3 octave increments) covering the frequency spectrum from 20 Hz to 20 kHz. They usually come with a calibrated electret microphone and a built-in pink noise generator for making acoustical measurements.

Rectified Voltage

A process wherein an alternating current signal is converted into a direct current amplitude modulated envelope representation of the source. Often, some smoothing is applied to this signal with a set of time constants referred to as "attack" and "decay." This signal is used in such devices as dynamic filters, companders, compressors, expanders, spectral enhancers, and is digitally simulated in some of the EIGHT/DC FORENSICS algorithms.

Residue

The residue of a filtered signal is the algebraic difference between the filter output and its signal input. EIGHT/DC FORENSICS allows you to hear the "residue" of two of its filters by enabling the "Keep Residue" function. The two filters that include this feature are the Continuous Noise Filter and the Harmonic Reject Filter. This feature has been included because in some cases, it may be useful as an aid to hear what you are filtering out of the signal source. This is particularly useful when adjusting the Harmonic Reject Filter when attempting to remove "Hum" or "Buzz" from a recording.

Resistor / Resistance

(R)

(Ohms) (Ω)

A basic electrical device that has electrical resistance, and is used to control the amount of current flow in a circuit. The unit of measurement for a resistor is the Ohm.

$$R = E / I$$

wherein:

R = Resistance in Ohms

E = Voltage in Volts

and

I = Current in Amperes

Resonance (Electrical)

A system in which two elements are operated in quadrature (each element operating at 1/4 the signals period) to produce a minimization or maximization of said signal.

$$Fr = 1/2 \pi (L \times C)^{1/2}$$

wherein:

Fr = Resonant Frequency in Hertz (Hz)

L = Inductance in Henries

C = Capacitance in Farads

Resolution

The minimum amplitude increment into which the A-D converter of a discrete time system can divide an analog signal. The resolution of EIGHT/DC FORENSICS is usually 16 bits, which is 1 part in 65,536. However, with the appropriate sound card, EIGHT/DC FORENSICS does support up to with up to 24-bit I/O resolution. Resolution can also refer to the minimum "time slice" into which a sampled data system is divided or displayed.

Reverse RIAA Curve

EIGHT/DC FORENSICS is equipped with a family of reverse RIAA curves, allowing you to use a standard RIAA phonograph pre-amplifier to perform your mastering of old acoustical and 78-RPM recordings. A straight reverse RIAA curve is supplied for acoustical recordings, and a number of reverse RIAA curves with varying values of turnover frequency are supplied for electrically recorded 78s. These reverse curves can be found as several of the equalizer factory presets. An RIAA curve encoded recording can be decoded using the Diamond Cut Virtual Phono Preamp.

Reverberation

The process whereby the acoustical reflections of a room or concert hall are reproduced artificially, with devices such as tapped delay lines working in conjunction with mixing and phase shifting devices or algorithms.

RIAA

Equalization Curve (Record Industry Association of America)

The RIAA Curve is an equalization frequency response contour which was utilized by some manufacturers of LP records after around 1955 and was made pretty much standard by 1960. It specifies three R*C time constants to be used by playback pre-amplifiers in order to invert the record cutter equalization. The three time constants and their corresponding breakpoint frequencies are as follows:

1. 3180 μ S (50 Hz)
2. 318 μ S (500 Hz) (turnover frequency)
3. 75 μ S (2120 Hz) (roll-off frequency)

This curve can be de-coded using the Diamond Cut Virtual Phono Preamplifier if you are using a flat hardware preamplifier.

RIAA / IEC Equalization Curve

The RIAA / IEC equalization curve is defined in terms of the same time constants as the RIAA curve, with one additional time constant added of 7960 μ S. This provides 3 dB of attenuation at 20 Hz rolling off at -6 dB / Octave thereafter. Below is a listing of all of the time constants associated with the RIAA / IEC Equalization Curve:

1. 7960 μ S (20 Hz) (low frequency rolloff)
2. 3180 μ S (50 Hz) (pull-out frequency)
3. 318 μ S (500 Hz) (turnover frequency)
4. 75 μ S (2120 Hz) (roll-off frequency)

Right Mouse Button

EIGHT/DC FORENSICS implements the following six functions with the right mouse button:

1. Play From Here

2. Preview From Here
3. Copy
4. Paste Over
5. Paste Insert
6. Mute
7. Zoom-In
8. Zoom-Out
9. Zoom-Out Full
10. Add a Marker
11. Label a Marker
12. Delete a Marker
13. Clear All Markers in Selected Area
14. Undo Last Edit
15. Snap Selection to Zero Crossing (Q)

Note: A pop-up menu will appear when using the right mouse button containing the above-mentioned items that you can select from when they are active. Items that are not active will be grayed out.

RMS (Root Mean Squared)

RMS is the square root of the average of the squared instantaneous values of a waveform taken over the waveforms time duration (sometimes referred to as the "effective" value or the "heating" effect value). In electrical terms, a.c. Voltages and currents can be described in terms of their RMS value; in acoustical terms, sound pressure (acoustomotive force) can be described in terms of its RMS value.

Roll-off

In the record industry, roll-off usually refers to the amount of attenuation in dB @ 10 kHz which must be applied during record playback in order to achieve a flat response on the high end of the audio spectrum. For example, the roll-off for the RIAA curve is -13.7 dB and -12 dB for the AES curve.

Roll-off Frequency

For a Low-pass filter or for an equalization curve (such as the RIAA curve), the upper cutoff frequency is sometimes referred to as the Roll-off Frequency.

RPM (Revolutions Per Minute)

Some common record speeds are 33.33 RPM for LPs, 45 RPM for records with the same name, 78.26* RPM for most so called electrically recorded lateral 78s (like Victor), 78.8 RPM for Edison Lateral's, 80 RPM for Edison Diamond Discs, and 160 RPM for Edison Cylinder recordings.

Additional speeds such as 16 RPM will occasionally be encountered. Here is a brief listing of some unusual speeds which may be encountered:

1. White Wax Cylinders (1888 - 1892): 100 RPM
2. Early Brown Wax Cylinders: 125 to 144 RPM
3. Brown Wax Cylinders (1892 - 1899): 125 RPM
4. Brown Wax Cylinders ("New Process" - 1900): 144 RPM
5. Edison Concert Cylinders: 100 RPM
6. Edison Gold Molded Cylinders: 160 RPM
7. Pre 1900 Berliner Discs: 57 to 72 RPM
8. Early Victor, Zonophone, Berliner: 71.3 RPM
9. 1908 to 1925 Victor Acoustics: 76.6 RPM
10. Electrical Era 78s: 78.26*
11. Edison Diamond Disc, Pathe', Brunswick's, and Okey: 80.0 RPM
12. Edison Electric Needle Type: 78.8 RPM
13. Some Early Microgroove LPs: 16.66 RPM

Fractional Speed (from 45 RPM) Change Speed ratio's are as follows:

1. 45 RPM to 78.26 RPM - Use +73.7 % speed change
2. 45 RPM to 78.8 RPM - Use +75.1 % speed change
3. 45 RPM to 80 RPM - Use +77.1 % speed change
4. Other values can be simply calculated by applying ratio-proportions.

*(actually 78.26086957 which is a 46:1 standard gear reduction from a 60 Hz Line operated Synchronous Motor turning at 3600 RPM))

Rumble

Rumble is a low frequency noise signal, typically below 50 Hz, which is often found on records. This phenomenon can be caused by seismic effects during the mastering process or during playback. On poor turntables or cutting lathes, it can also be produced by irregularities in the main thrust-bearing race. To attenuate turntable rumble

using EIGHT/DC FORENSICS, use the High-pass Filter. Start with settings of 60 Hz and 18 dB / Octave (or steeper), and adjust the frequency upwards or downwards until you are satisfied with the results.

Sample Rate

The rate at which an analog signal is converted to discrete numbers by an A-D converter. For audio systems, sample rate is expressed in kHz. EIGHT/DC FORENSICS supports any number of standard sample rates including:

1. 11.025 kHz
2. 22.05 kHz
3. 44.1 kHz
4. 48.00 kHz
5. 88.2 kHz
6. 96.00 kHz
7. 192.00 kHz
8. Your entered choice of a numerical value up to 192.00 kHz

If your sound card supports intermediate sampling rates, you can also enter the numeric value of any sample rate you desire, between 8 kHz to 192 kHz for recording purposes.

Sampling Theorem

In a sampled data system (like the environment in which your EIGHT/DC FORENSICS program is operating), sampling theorem tells us that regularly spaced sampling must occur at least at the Nyquist rate, which is twice the frequency of the highest frequency signal or noise component that is expected to be resolvable by the system (without aliases). In other words, in a system expected to exhibit a frequency response up to 20 kHz, the minimum sample rate will have to be 40 kHz. Because it is impossible to construct an ideal Low-pass filter, the sampling rate will have to be somewhat larger than 2X the desired maximum frequency response value. In practice, a 44.1 kHz sampling rate is generally used in 20 kHz frequency response audio applications (although sometimes 48 kHz and 96 kHz are also used).

Shielded Cables

Shielded cables are special cables which are designed to minimize stray noise fields (particularly E fields) from entering an audio system through the interconnection wiring

from component to component due to extraneous sources. Most often shielded cables are of the co-axial type so that loop area is also minimized, resulting in a minimization of "H" field pickup. However, some systems use a balanced pair of shielded wires which further minimizes pickup, provided the appropriate terminating transformers or differential amplifiers & line drivers are used on each end of the cable.

SID

Slewing-Induced Inter-modulation Distortion. This is similar to TIM. See TIM for details.

Signal-to-Noise Ratio

The ratio of signal-to-noise (Voltage, current, or acoustical sound pressure level) that is expressed in dB. Signal-to-Noise ratio in dB = $20 \log (\text{signal} / \text{noise})$.

SINAD (Signal, Noise And Distortion)

Essentially, SINAD is the reciprocal value of a THD measurement and is expressed in dB.

$$\text{SINAD} = 20 \log (\text{signal rms value} + \text{noise \& distortion}) / (\text{rms value of noise \& distortion})$$

Single-Ended Noise Reduction

Single-Ended Noise Reduction are the processes wherein noise is removed from an un-encoded audio signal. Algorithms like the Impulse, Continuous, Dynamic, Harmonic Reject, and Notch filters are examples of Single-Ended Noise Reduction tools.

Slew Rate

Slew rate is the maximum dv / dt (and sometimes the maximum di / dt) that an audio system component can react to. Parasitic Miller capacitances in operational amplifiers are generally the cause of this parametric limitation. When an audio component is confronted by a signal exceeding its maximum slew rate capability, the system reverts to slew rate-limited mode of operation. During this time interval, all audio information is lost resulting in severe distortion of the applied input signal.

Slope

In the context of EIGHT/DC FORENSICS and audio filter terminology, slope is the linear rate of change of amplitude vs.

frequency of a filter past its corner frequency. This is expressed in dB / Octave or dB / Decade. $6 \text{ dB / Octave} = 20 \text{ dB / Decade}$, $12 \text{ dB / Octave} = 40 \text{ dB / Decade}$, etc.

Slot Filter

A slot filter is the compliment to the "notch" filter. It is a variable narrow Band-pass filter; capable of greater selectivity than a typical Band-pass filter. It is often used in Forensics work for isolating particular sounds like the ringing of a telephone on a recording in a crowded noisy bar situation, or anything similar. By allowing only a very narrow "slot" of frequencies through the system, one can observe the "slotted-band" with a much improved signal to noise ratio compared to the wideband signal. The EIGHT/DC FORENSICS slot filter can be found under the Notch filter and is activated by checking the appropriate box. Multiple slot filters can be run via the Multi-Filter. If the slots that are desired are harmonically related, you could use the Harmonic reject filter in "keep-residue" mode to produce up to 500 slots in one pass.

Sone

A Sone is a unit of measurement for sound loudness. A simple tone of a frequency of 1 kHz and at a level 40 decibels above a listener's threshold of perception represents a loudness of 1 Sone. A loudness of any sound which is judged by a listener to be "n" times greater than that of the 1 Sone tone is defined as "n" Sones.

Sound Level

Sound Level is a weighted sound pressure level obtained by the use of a metering system and any of three weighting standards as established in the American National Standard Specification for General Purpose Sound Level Meters. The reference pressure is 2×10^{-5} Newton per meter 2 . The two most common standards are the "A" and the "C" weighting factors. The "A" weighting characteristic responds mostly to frequencies in the area of the greatest sensitivity of the human ear in the 500 to 10,000 Hz range. The "C" weighting characteristic is nearly uniform over most of the audio spectrum.

The 0 dB reference sound pressure level (SPL) for a sound level meter is 0.0002 microbars using a simple tone of 1000 Hz.

For a chart showing several common sound sources and their Acoustic Power and Sound Power levels (from 10 meters), simply turn to the ***Charts, Graphs, and Other Useful Info*** section.

Sound Wave Velocity

Sound Wave Velocity in air as a function of temperature is given by the following:

$$c = 33,100 (1 + 0.00366t)^{1/2}$$

wherein:

c = Sound Wave Velocity in air in centimeters per second

and

t = temperature in degrees centigrade

Therefore at 70 degrees C, sound will travel at 37,098.6 centimeters per second, or around 830 miles per hour.

Sound Wavelength

The Wavelength of a sound wave is given by the following equation:

$$\lambda = c / f$$

wherein:

λ (lambda) = wavelength in centimeters

and

c = Sound Wave Velocity

and

f = frequency in Hz (cycles per second)

S/PDIF

S/PDIF (Sony / Philips Digital Interface Format) is a serial digital signal format used to connect digital audio devices together. It is a subset of the IEC 60958 (or AES/EBU) standard. It is electrically characterized as a coaxial system terminated with 75 Ohms on each end and typically using RCA (orange in color) or BNC connectors.

Speech Filter

A filter which typically has a Band-pass only between the frequencies of 300 Hz to 3 kHz, and which is used to improve the basic

intelligibility of speech. Often, this type of filter uses slopes of -12 dB / Octave.

This characteristic can be replicated with the Band-pass filter. An alternative speech filter that is sometimes useful is called the Steep Slope Speech filter. Its response is 250 Hz to 3.5 kHz with a slope of 18 dB / Octave.

Speed

See RPM

Spectrograph

A Spectrograph is a system for presenting audio data in a graphical form and is a special case of a spectrum analyzer coupled to an oscilloscope. The horizontal (X) axis represents time, the vertical (Y) axis plots frequency and the gray scale brightness (or color) (Z axis) represents the signal intensity. Audio Spectrographs are used forensically for spectrographic voice recognition (sometimes referred to as voiceprints) and acoustical analysis applications. The software includes a Spectrograph feature, which is found under the Forensics menu.

Spectrum

Spectrum is band or range of frequencies as in the audio spectrum, the light spectrum, or the electromagnetic spectrum.

Spectrum Analyzer

A spectrum analyzer is a device for analyzing and displaying the Amplitude versus Frequency characteristic of a portion of a spectrum. They fall into two general categories:

1. Swept Band-pass Filter (a serial process of analysis)
2. Real Time Analyzer (a parallel process of analysis)

Spectral Enhancer

An electronic device which is used to expand the dynamic range of the upper and/or the lower octaves of the audio frequency spectrum, leaving the mid-band portion of the spectrum unprocessed. This has the effect of increasing the "definition" of a recording without continuously amplifying hiss and rumble which may be present on the source material. It is a form of dynamic filter which uses the principle of "upward expansion" to improve dynamic range. The Dynamic Noise Filter contains a

Spectral Enhancer mode of operation, which can be enabled.

Spectral Subtraction

Spectral subtraction is a method for reducing the unwanted noises associated with a recording using the amplitude elements of the frequency domain information contained in a .wav file. A noise fingerprint is taken, and this frequency domain information is then subtracted from the rest of the file and then re-converted back to the time domain. This technique is primarily used in forensics audio applications, and can be found as one of the options in the Continuous Noise filter.

Stroboscope

A device which indicates the RPM speed of a turntable by creating an optical illusion of the slowing-down, freezing, or speeding-up of a pattern when illuminated by a pulsating light source operating at a known frequency. You can create your own stroboscope disc by dividing a circle evenly into black and white segments. Use the following formulae to calculate the number of segments required per 360 degrees (1 rotation of the disc) into which the disc must be marked:

60 Hz power systems: # of segments = $7,200 / \text{RPM}^*$

50 Hz power systems: # of segments = $6,000 / \text{RPM}^*$

For example, assume that you want to construct a strobe for use in the United States where the power system operates at 60 Hz in frequency. We want to design it "to freeze" at 78.2 RPM. $7,200 / 78.2 = 92.07$. Round the number to 92 segments. Divide your circle into 92 evenly spaced segments, and voila, you have your strobe. Because of the rounding error, the strobe you constructed will be in error by 0.08 %. Your strobe will have to be used under a fluorescent or neon light connected to the power line in order to function. Incandescent lamps will not work because of the long thermal time constant of their filaments.

For a chart that will help you create your own strobe using common line frequencies and RPM values, go to our *Charts, Graphs, and Other Useful Info* section.

Note: EIGHT/DC FORENSICS provides two bitmaps that you can print and use as phonograph strobes covering the important

speeds. These can be found in the Diamond Cut Directory at "Strobe50Hz.wmf" and "Strobe60Hz.wmf".

Sub-harmonics

Sub-harmonics are fractional multiples of a fundamental frequency. The Diamond Cut Sub-harmonic synthesizer is capable of creating certain signals of this nature in the bass end (below 75 Hz) of the audio spectrum.

Square Wave

A square wave is a waveform consisting of a fundamental frequency and the sum of all of the odd harmonic components of that fundamental frequency on the frequency spectrum up to an infinite number of harmonics. An ideal square wave contains approximately 43% Total Harmonic Distortion (THD).

Styli

The following is a listing of some of the more common record types and the styli that they require:

- A. Modern LPs: 0.7 mil elliptical
- B. Early LPs: 1.5 mil truncated elliptical
- C. Transcription Recordings: 2.3 mil truncated elliptical
- D. Narrow Groove 78s such as Polydor: 2.4 mil truncated elliptical
- E. Late 1930's Lateral 78 RPM Discs: 2.8 mil truncated elliptical
- F. Standard Groove 78 RPM Discs: 3.0 mil truncated elliptical
- G. Pre-1935 Lateral Cut Electrical 78s: 3.3 mil truncated elliptical
- H. 1931 to 1935 RCA Pre-Grooved Home Recordings: 5.0 mil spherical
- I. Edison 80 RPM Diamond Discs: 3.7 mil spherical or non-truncated conical
- J. Edison Blue Amberol Cylinders: 3.7 to 4.2 mil non-truncated spherical
- K. Wide Groove Acoustical 78 Lateral Disc: 3.8 mil truncated elliptical
- L. Edison Wax Amberol Cylinders: 4.2 mil Spherical

- M. Edison White Wax, Brown Wax, Concert, and Gold Molded Cylinders: 7.4 mil Spherical
- N. Pathé 78s: 3.7 mil truncated conical
- O. Metal Stampers: Bi-radial of appropriate dimensions *
- P. Late 16 inch transcription discs: 2.0 mil truncated elliptical
- Q. Very early acoustical lateral cut discs: 4.0 truncated elliptical
- R. Etched-label Pathé up to 14 inches in diameter: 8.0 mil spherical
- S. Etched-label Pathé greater than 14 inches in diameter: 16.0 mil spherical
- T. Acetate and aluminum "instantaneous" discs: 6.0 mil elliptical or truncated elliptical
- U. Very Late 16 inch Transcription Discs: 2.0 x 6.0 elliptical
- V. Wagner-Nichols Records: 0.5 mil truncated elliptical
- W. Shallow Groove electrical recordings 2.0 truncated elliptical
- X. Transcription. 1930's and 1940's 16 inch Acetates: 2.6 truncated elliptical
- Y. General purpose 78 RPM stylus: 3.0 mil truncated elliptical

* Note: When stampers are played on a conventional turntable equipped with a Bi-radial stylus, you will need to use the File Reversal feature so that it can be converted to forward play.

Tape Head

A tape head is an electromagnetic device used in a tape recorder to apply and read magnetic signals onto (and from) magnetic tape media. It consists of a coil mounted on a magnetic structure having a "gap" where the tape comes in contact. The tape head gap width in conjunction with the magnetic particle size on the tape media determines the frequency response of the system. This process follows Faradays Law of electromagnetic induction:

$$V_{\text{head}} = - N d\phi / dt$$

wherein

V_{head} = Tape Head Voltage

N = Number of turns of wire on the tape head structure

$d\phi / dt$ = The time derivative of magnetic flux

Since $d\phi / dt$ is proportional to and increases with frequency, this tape head Voltage signal increases at a rate of 6 dB / Octave and must be compensated for to prevent tape saturation. The process to provide this compensation is called tape equalization.

Tape Head Gap

The Tape head gap is the discontinuity in the magnetic pathway (or circuit) formed in the tape head structure which runs perpendicular the direction of the tape movement. Ideally, the playback head gap should be smaller than one half the Lambda (or wavelength as it relates to the speed of the tape) of the highest frequency signal to be recorded. The following is a listing of typical Tape Head Gaps by Tape Head type:

Playback Heads: 1 to 5 microns (micro-meters)

Recording Heads: 3 to 13 microns (micro-meters)

Erasur Heads: 25 to 150 microns (micro-meters)

Tape Recorder Speeds

See IPS

THD (Total Harmonic Distortion)

THD is a figure of merit as to how much non-linearity a system is imposing upon an audio conduit. The Spectrum Analyzer has the ability to measure the THD of an item under test when used in conjunction with either the Make Waves generator or an external hardware equivalent. %THD = Amplitude of the Harmonic Content of a signal / Amplitude of the Fundamental Component.

When using hardware to make this measurement on very low distortion equipment, it is necessary to account for the Generator THD. Therefore, one must measure the test Generators THD at each test frequency as well as the System Measured THD (of the whole system). Then, one must apply the following equation:

$$\text{Actual THD} = (((\text{System Measured THD})^2 - (\text{Generator THD})^2)^{1/2})$$

THD + N (Total Harmonic Noise plus Noise)

This is what is actually measured by most THD meters, including not only the harmonic distortion created by the device being tested, but its noise as well.

Thermal Noise (Floor)

Any electrical conductor produces a random noise Voltage as long as exists above 0 degrees K and/or has an electrical resistance greater than zero Ohms. The following formulae can be used to calculate the Root Mean Square value of the thermal noise Voltage of a terminating or source resistance:

$$E = (4RkT \times \Delta f)^{1/2} \text{ or } E = \sqrt{(4RkT \times \Delta f)}$$

Wherein:

R = Resistive Component in Ohms (Ω)

k = Boltzmann's Constant = 1.38×10^{-23}

Joules / Kelvin (1 Joule = 1 Watt x Second)

T = Absolute or Thermodynamic

Temperature in degrees Kelvin

Δf = Bandwidth of the system in Hertz (Hz)

E = Root Mean Square (RMS) Noise

Voltage

T (in degrees Kelvin) = Temperature in

Degrees C + 273.15

Example: Assume an audio mixer/microphone preamplifier is terminated with a 50K Ohm Resistance. It is operating at 40 degrees C internally, has a 60 dB Voltage Gain and exhibits a usable flat response from 20 Hz to 20 kHz. What is the RMS noise floor of the output of the mixer?

$$60 \text{ dB Voltage Gain} = 1000 : 1 = 1000$$

$$E_{\text{out}} = (4RkT \times \Delta f)^{1/2} \times 60 \text{ dB} = ((4 \times 50,000) \times (1.38 \times 10^{-23}) \times (30 + 273.15) \times (20,000 - 20))^{1/2} \times (1000)$$

$$E_{\text{out}} = 0.004 \text{ Volts RMS} = 4.0 \text{ Millivolts RMS}$$

It is important to note that this is the Noise Floor for this system and can't be made to be any quieter than this number unless

special techniques are employed. Cryogenic cooling techniques are sometimes used on first-stage amplification devices of specialized amplifiers to improve the noise performance of highly sensitive systems.

TIM (Transient Inter-modulation Distortion)

This is a form of distortion that occurs when a system enters into a "slew limit" mode of operation during a fast audio transient. The result is the loss of all sonic information during the "slew interval." It is usually the result of poorly designed amplifiers having slow error amplifiers or insufficient high frequency current drive provided to power amplifier output transistors. This type of distortion is sometimes referred to as SID (slewing induced inter-modulation distortion) or DIM (Dynamic inter-modulation distortion).

Time Constant

Time constants are exponential amplitude vs. time functions, which are realized with resistors and capacitors, or resistors and inductors.

$$\text{Tau} = R \times C \text{ for Resistor/Capacitor circuits}$$

or

$$\text{Tau} = L / R \text{ for Resistor/Inductor circuits}$$

wherein:

Tau = time constant in seconds

R = resistance in Ohms

C = capacitance in Farads

and

L = inductance in Henries

The relationship between a simple first order filters corner frequency (F_c) and time constant is as follows:

$$F_c = 1 / (2 \times \pi \times \text{Tau})$$

Note that the higher the value of time constant, the lower the corner frequency created. Some common time constants found in audio applications are as follows:

25 μSec - Dolby based FM de-emphasis

70 μ Sec - Type 1 (Normal Bias) Cassette Tape Eq.

75 μ Sec - Standard FM Broadcast de-emphasis

120 μ Sec - Type 2 (High Bias) Cassette Tape Eq.

Additional audio time constants can be found under RIAA and NAB in this glossary.

Time Derivative

This is the instantaneous rate of change of a parameter (such as Voltage, amplitude, or sound pressure level) with respect to time. (i.e. dV/dt , dP/dt , etc.)

Tracer Technologies

Tracer Technologies is one of our software distributors. Here is some contact information for Tracer:

Tracer Technologies
P.O. Box 189
Windsor, PA
17366

717 764 9240
fax 764 9254

www.tracertek.com
info@tracertek.com

Tremolo

Tremolo is the amplitude modulation of a musical note. For example, the tremolo control on a guitar amplifier modulates the gain of its signal by way of a low frequency sine wave oscillator in the 1 to 10 Hz range. A Tremolo effect can be viewed on the Diamond Cut Spectrogram.

Transformer

An alternating current device used to impedance match transducers and electronic circuits to one another. Sometimes, these devices are used with a unity turns ratio to provide isolation from one circuit to another rather than to impedance match the two. This is useful in audio applications when it is necessary to break a ground loop source of noise in a system.

Triode

A Triode is an electron tube (or valve) containing three elements. They consist of an anode, cathode, and a control grid. Small

changes in grid Voltage produce large changes in values of current in the plate circuit (the ratio of delta plate current to delta grid Voltage is its gain in transconductance or μ .) They are most commonly used in audio pre-amplifier, and other low-level applications. Typical triodes found in audio applications include the 12AX7 and 6SL7 high μ (gain), and the 12AU7 and 6SN7 medium μ devices. All of the devices listed are "dual" (two in one envelope).

Tube

See Electron Tube and/or Valve

Turnover Frequency

The frequency in a phonograph equalization curve below which the master was recorded with the cutting head operating in constant displacement mode rather than in constant velocity mode. This is used to limit the excursions of the cutting stylus so that bass notes do not cause the cutting stylus to break through to the adjoining groove wall. For a chart listing the most common Turnover Frequencies utilized by 78 RPM records, refer to our *Charts, Graphs, and Other Useful Info* section.

Valve

The British term for an electron tube. It arises out of the valve like effect that a grid has on the flow of electrons between the devices cathode and anode (or plate). Also, refer to Electron Tube.

Vector Quantity

Any physical quantity, like the displacement of a record stylus, whose specification involves both magnitude and direction and which obeys the parallelogram law of addition. The Diamond Cut X-Y display provides a visual indication of the Vector Quantity consisting of one audio channel plotted against the other.

Vertical Cut

(Also known as "Hill and Dale")

A record recording technique in which the groove modulation (undulations) occur in an up-and-down direction as opposed to side-to-side. This technique was used by Thomas Edison in his original invention of the phonograph, and was maintained as the recording method used by his companies cylinders and Diamond Discs.

Vibrato

Vibrato is the frequency modulation of a musical note. It results in what is perceived as a pulsating change in timbre. A typical vocal embellishment of this type created by singers will occur at a rate of change in frequency occurring around 6 to 8 Hz. This effect can be viewed on the Diamond Cut Spectrogram.

Voice Fundamental Frequency (F0)

The fundamental frequency of a human voice is sometimes referred to as F0. For adult females, that range runs from 165 to 255 Hz and for adult males the range runs from 85 to 180 Hz. Speech articulation uniqueness is the result of the ratio of formant frequencies F1 through F4.

Vocal Ranges

Human Vocal ranges (relative to middle C = C4 = 262 Hz) can be broken down from lowest to highest as follows:

Bass: E2 to E4

Baritone: F2 to F4

Tenor: C3 to C5

Contralto: F3 to F5

Mezzo-Soprano: A3 to A5

Soprano: C4 to C6

The notes in these ranges can be created via presets found in the Make Waves Generator.

Voice Frequency Range

The fundamental component frequency range for mature adult, healthy human voices are as follows:

Female: 165 to 255 Hz (Harmonics up to 10 kHz)

Male: 85 to 155 Hz (Harmonics up to 8 kHz)

Volt

(V)

(Voltage)

The unit of measurement of electrical potential difference (or electromotive force) equal to the difference in potential which occurs in a conductor which is carrying 1 Ampere, and the power being dissipated in the conductor is 1 Watt, with the resistance of the conductor being 1 Ohm.

Vorbis (Ogg Vorbis)

Vorbis is a lossy audio compression technique (codec) that is supported by your Diamond Cut software. It is especially useful in low bit rate audio applications (less

than 128 kbit/sec). Its file extensions are .ogg and .oga.

VOX

VOX is an acronym for "Voice Operated Transmit" which is derived from the half-duplex two-way radio field. In DC terms, it applies to the automatic activation of the recording function when a signal is detected by the system. The VOX system also ceases recording after the signal vanishes for more than a user settable period of time.

Watt

(P)

(Power)

A measurement unit of electrical power equal to the ability to do work at the rate of 1 Joule per second.

$P = V \times I$ wherein P = power in Watts, V = Voltage in Volts, and I = current in Amperes.

Wave file (.wav)

The primary and native pcm (pulse code modulation) sound file format that DC8 & EIGHT/DC FORENSICS supports. This (.wav) is the standard Windows file format.

Wet

The signal output of a special effects generator (such as the Reverb), which contains the modified (processed) signal. "Wet" refers to the effects signal alone. The non-processed signal from such a generator is referred to as "dry." As with most special effect generators, the Reverb has an output mix control which allows you to transfer a signal from the effects generator, which ranges from completely dry to completely wet (no source signal), or to some mixture in between.

White Noise

White Noise is random noise that is characterized as containing equal energy per unit frequency (Hertz). White Noise is sometimes referred to as Johnson, shot, or thermal noise. White noise derives its name from the analogous definition of white light. Audio white noise can be created using the Make Waves function using the function by that same name. White Noise can be converted to Pink Noise with a Multifilter preset. Please refer to the Pink Noise section of this Glossary for more information..

Window Weighting

Window weighting is a concept that pertains to systems, which involve fast Fourier transforms (FFT's). Signals, which are observed for finite intervals of time, may contain distorted spectral data in the transform due to the ringing of the $\text{Sin}(f)/f$ spectral peaks of a rectangular window. This distortion is minimized by the use of a window-weighting function, which is applied before the DFT is performed. The window weighting functions used in the FFT based EIGHT/DC FORENSICS algorithms is proprietary.

.wma File Format (Windows Media Audio)

Windows Media Audio format is Microsoft's implementation of compressed audio files and uses the file extension .wma. You can create files of this type using the Diamond Cut "Save As" command found under the File Menu. Both lossy and lossless compression routines are available.

Wow

A slow periodic change in the pitch or low frequency flutter which may be present on phonograph, tape, or soundtrack recordings due to a non uniform velocity of the recording medium. Wow is generally a frequency modulating effect that occurs at a deviation rate between 0.5 to 6 Hz. The

Wow could have been introduced in the recording process, the playback process, or a combination of both. Wow found on record recordings is usually caused by a non-concentric spindle hole. Wow found on tape recordings is generally caused by warped take-up or supply reels. EIGHT/DC FORENSICS is not capable of correcting audio problems of this nature at this point in time.

Wow and Flutter

Wow and flutter is the combined FM effect of both mentioned parameters. The frequency spectrum in which this rate of frequency deviation is made is in the spectrum that exists between 0.5 to 250 Hz.

X-Axis

This is the horizontal axis of a graph. In EIGHT/DC FORENSICS, it contains the time information for your .wav file that is divided up into ten equally spaced grids.

Y-Axis

This is the vertical axis of a graph. In EIGHT/DC FORENSICS, it contains the amplitude information for your .wav file that is divided up into four equally spaced grids.

Z-Axis

This represents the gray scale intensity or chroma modulation level that you find in the spectrogram display.

Charts, Graphs and Other Info

Additional Technical Information

We've recently added a section of our web site with specific Application Notes making use of our products. These App Notes provide stimulating specific information on uses for our programs. These currently include Using DC EIGHT or DC Forensics as an Audio Signal Generator, Testing your Sound Card's Performance, Electric Network Frequency Analysis, and more. You can find these at:

<http://www.diamondcut.com/AppNotes.htm>

The Engineers at Diamond Cut Productions also maintain a very active BBS / Forum where users intermingle with the programmers and ask questions about the products. You may find a visit here to be very informative.

<http://www.diamondcut.com/vforum/index.php>

A Presets Sharing Forum is available for your use at:

<http://www.diamondcut.com/vforum/forumdisplay.php?f=13>

Download Demo's and Patches can be found at the following site:

<http://www.diamondcut.com/support.htm>

The Diamond Cut Productions, Inc Home Page URL is:

<http://www.diamondcut.com/>

Attenuation Chart

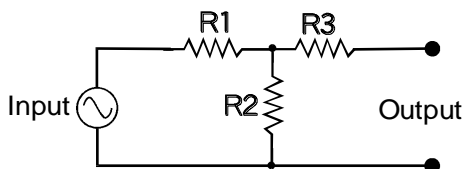
Here is a table of resistance multipliers for a symmetrical (equal input and output impedance) "T" Pad attenuator:

R1 = Attenuator Input Resistor

R2 = Attenuator Shunt Resistor

R3 = Attenuator Output Resistor

(Note: R1 = R3 & the output impedance of the Input Source and the input impedance of the Output Load circuit must also be equal to R1 and R3)



Attenuation (dB)	R1 & R3 (normalized Ohms)	R2 (normalized Ohms)
0.0	0.000000	infinite
0.5	0.028775	17.362
1.0	0.057501	8.6668
2.0	0.114620	4.3048
3.0	0.171000	2.8385
4.0	0.226270	2.0966
5.0	0.280130	1.6448
6.0	0.332280	1.3386
7.0	0.382480	1.1160
8.0	0.430510	0.94617
9.0	0.476220	0.81183
10.0	0.519490	0.70273
20.0	0.818180	0.20202
40.0	0.980198	0.020002
60.0	0.998000	0.0020000
80.0	0.999800	0.00020000
100.0	1.000000	0.000020000

To use this table, multiply the input (or output) impedance of your circuit by the numbers associated with the attenuation that you desire. Remember, this table of values requires that the input terminating impedance and the output terminating impedance of the circuits on each side of the attenuator be present and of the same value. To obtain values of attenuation that are not in this table, merely cascade "T" sections adding up to the value (in dB) which you desire. For example, to achieve 23 dB, cascade a 20 dB section with a 3 dB section.

Audio Connection Standards

1. **Balanced "XLR" Standard**

- A. Pin # 1 = Shield (Common)
- B. Pin # 2 = + (Hot)
- C. Pin # 3 = - (Cold)

2. **1/4 inch Stereo Phone Plug (TRS) for Balanced Audio Circuits**

- A. Tip = + (Hot)
- B. Ring = - (Cold)
- C. Sleeve = Shield (Common)

3. **1/4 inch Mono Phone Plug (TR) for Unbalanced Audio Circuits**

- A. Tip = + (Hot)
- B. Sleeve = - (Shield)

4. **RCA / Phono Plug**

- A. Tip = + (Hot)
- B. Sleeve = - (Shield)

5. **Amphenol 3 Pin Balanced Microphone Connector**

- A. Pin #1 = Shield (Common)
- B. Pin #2 = + (Hot)
- C. Pin #3 = - (Cold)

6. **Amphenol 4 Pin Microphone Connector (Balanced and Unbalanced)**

- A. Pin #1 = Shield (Common)
- B. Pin #2 = + (Hot) Unbalanced (Note: Unbalanced output is with respect to Shield)
- C. Pin #3 = + (Hot) Balanced
- D. Pin #4 = - (Cold) Balanced

7. DIN5 Pin Connector (Tape Deck I / O Connector)

- A. Pin #1 = Right Channel Record Input
- B. Pin #2 = Shield (Common)
- C. Pin #3 = Right Channel Playback Output
- D. Pin #4 = Left Channel Record Input
- E. Pin #5 = Left Channel Playback Output

8. 1 / 8 inch Mono Phone Plug (TR)

- A. Tip = + (Hot)
- B. Sleeve = - (Shield)

9. 1 / 8 inch Stereo Phone Plug {The type used on most Sound Cards}

- A. Tip = Left Channel + (Hot)
- B. Ring = Right Channel - (Hot)
- C. Sleeve = Shield (Common)

10. 1 / 8 inch Microphone Input Plug (TRS) {The type used on most Sound Cards}

- A. Tip = Signal + (Hot)
- B. Ring = Phantom Power (~3 to 4 Volts @ ~ 0.75 to 1.5 mA)
- C. Sleeve = Shield (Common)

11. Modular Telephone Jack Wiring (- 48 Volt, 4 terminal, 2 line system / United States)

- A. Red* or Blue or Blue with White Stripe = Line #1 (negative) (Hot)
 - B. Green* or White or White with Blue Stripe = Line #1 (positive) (Common)
 - C. Yellow* or Orange or Orange with White Stripe = Line #2 - (Hot)
 - D. Black* or White or White with Orange Stripe = Line #2 + (Common)
- * Denotes the most standard color code

12. Standard USB (Universal Serial Bus) Pinout and Color Code

- A. Pin #1 = Vbus (5 Volts @ 500 mA) (Red Wire)
- B. Pin #4 = Vbus Ground Return Conductor (Black Wire)
- C. Pin #3 = Data + (Green Wire)
- D. Pin #2 = Data - (White Wire)

Note: Data + and Data – Conductors are a Twisted Pair

Audio Frequency Spectrum

The following is a listing of some common audio sources and the portion of the audio spectrum that they typically occupy, including their harmonics:

Audio Source	Fundamentals + Harmonics	Fundamental Only
Accordion	130 Hz to 15 kHz	130 Hz to 1.8 kHz
Bass Drum	50 Hz to 5 kHz	----
Bassoon	65 Hz to 9 kHz	65 to 650 Hz
Bass Tuba	40 Hz to 7 kHz	40 Hz to 375 Hz
Bass Viola	45 Hz to 8 kHz	45 to 300 Hz
Cello	70 Hz to 14 kHz	70 Hz to 900 Hz
Clarinet (Soprano)	150 Hz to 14 kHz	150 Hz to 1.7 kHz
Cymbals (14 inch)	300 Hz to 17 kHz	----
Female Speech*	165 Hz to 10 kHz	165 to 255 Hz
Flute	250 Hz to 14 kHz	250 Hz to 2.5 kHz
Foot Steps	80 Hz to 15 kHz	----
French Horn	70 Hz to 6 kHz	70 to 825 Hz
Hand Clapping	100 Hz to 15 kHz	----
Harmonica	450 Hz to 15 kHz	450 Hz to 1.3 kHz
Jingling Keys	1.5 kHz to 14 kHz	----
Male Speech*	85 Hz to 8 kHz	85 to 180 Hz
Oboe	250 Hz to 15 kHz	250 Hz to 1.7 kHz
Piano	30 Hz to 6 kHz	30 Hz to 4.2 kHz
Piccolo	500 Hz to 15 kHz	500 Hz to 3.8 kHz
Pipe Organ	16 – 32 Hz to 15 kHz	16 – 32 Hz to 8 kHz
Room Noise	30 Hz to 18 kHz	----
Snare Drum	80 Hz to 15 kHz	----

Timpani Drums	50 Hz to 4.5 kHz	----
Trombone	80 Hz to 8 kHz	80 Hz to 500 Hz
Trumpet	180 Hz to 9 kHz	180 Hz to 900 Hz
Violin	190 Hz to 15 kHz	190 Hz to 3 kHz

*Note: Frequency response is specified for mature & healthy adults

Compact Discs

There are a number of standards for data contained on Compact Discs. They are as follows:

Type	Application	Comments
Red Book	CD Audio / Compact Disc	PCM, 44.1 kHz sampling, 16 bit x 2 channels, 588 bits/frame, 192 bits/frame for the audio stream.
Yellow Book	Computer Data	Data structure based on ISO 9660
White Book	Video CD	MPEG audio/video track encoding
Blue Book	Enhanced Music CD (Audio + Data)	Structure similar to ISO 9660
Orange Book	CD-MO, CD-R & CD-RW	Magneto Optical / CD Write Once / CD Re-Writable
Photo CD Book	Photographs	Based on CD-I Bridge spec.
Multi-session CD	Multiple Session not recordable	Data structure based on ISO 9660

Decibels

The following table shows the relationship between Voltage, Current, and Power ratios and Decibels:

Numerical Ratio	Voltage or Current Ratio in dB	Power Ratio in dB
1 : 1	0	0
2 : 1	6.0	3.0
3 : 1	9.5	4.8
4 : 1	12.0	6.0
5 : 1	14.0	7.0
6 : 1	15.6	7.8
7 : 1	16.9	8.5
8 : 1	18.1	9.0
9 : 1	19.1	9.5
10 : 1	20	10
100 : 1	40	20
1,000 : 1	60	30
10,000 : 1	80	40
100,000 : 1	100	50
1,000,000 : 1	120	60
10,000,000 : 1	140	70
100,000,000 : 1	160	80
1,000,000,000 : 1	180	90
10,000,000,000 : 1	200	100

Dial Tone Phone Frequency Chart

The Spectrum Analyzer can be used to detect and identify a telephone number dialed when used in conjunction with this chart. Also included are the letters A, B, C, and D, which was used in the US military's Autovon phone system.

1	2	3	A	697 (687–708)
4	5	6	B	770 (759–782)
7	8	9	C	852 (839–865)
*	0	#	D	941 (927–955)
1209 (1191–1227)	1336 (1316–1356)	1477 (1455–1499)	1633 (1609–1658)	Frequency (Hz)

Note 1: The tolerance for these frequencies is +/- 1.5 %, and the range of which is shown in parenthesis. The highest frequency must also be as loud as the lowest frequency, or as much as 4 dB louder (this difference in level is referred to as “twist.”)

Note 2: Call waiting tone in the US is 440 Hz.

Note 3: Caller ID on call waiting in the US is 2130 + 2750 Hz.

Worldwide Dial Tone Frequencies

Country	Freq.1	Freq. 2	Country	Freq.1	Freq.2
Belgium	450 Hz	Not Used	Singapore	270 Hz	320 Hz
France	440 Hz	Not Used	South Korea	350 Hz	440 Hz
Germany	425 Hz	Not Used	Sweden	425 Hz	Not Used
Israel	400 Hz	Not Used	Switzerland	425 Hz	Not Used
Italy	425 Hz	Not Used	Taiwan	350 Hz	440 Hz
Japan	400 Hz	Not Used	U.S.	350 Hz	440 Hz
Netherlands	150 Hz	450 Hz	U.K.	350 Hz	Not Used
Norway	425 Hz	Not Used			

*Note: These can be useful as a reference frequency when measuring DTMF signals

Dynamic Range

Below is a table of common values of audio system resolution and their associated dynamic ranges:

Number of Bits of Resolution	Theoretical Maximum Dynamic Range
8 bits	48 dB
16 bits	96 dB
20 bits	120 dB
24 bits	144 dB

Equalization Curves (Phonographic)

The following is a listing of turnover frequencies and roll-off attenuation values for the various equalization curves that were used for playback by the phonographic industry (many of which can be found as presets in the Diamond Cut Virtual Phono Preamp {VPA}):

Equalization Curve	Turnover Frequency	Roll-off dB @ 10 kHz
AES	400 Hz	- 12 dB
Columbia LP	300 Hz	- 16 dB
EMI LP	500 Hz	- 10.5 dB
ffrr (1949)	250 Hz	- 5 dB
ffrr (1951)	300 Hz	- 14 dB
ffrr (1953)	450 Hz	- 11 dB
NAB	500 Hz	- 16 dB
NARTB	500 Hz	- 12 dB
RCA Early Orthophonic	500 Hz	- 11 dB
RCA New Orthophonic	500 Hz	- 13.7 dB
RIAA	500 Hz	-13.7 dB

RIAA Curve Table of Values

Frequency in Hz	Level in dB referenced to 0 dB @ 1 kHz*	Frequency in Hz	Level in dB referenced to 0 dB @ 1 kHz*
20	+ 19.3	800	+ 0.7
30	+ 18.6	1,000	0.0 *
40	+ 17.8	1,500	- 1.4
50	+ 17.0	2,000	- 2.6
60	+ 16.1	3,000	- 4.8
80	+ 14.5	4,000	- 6.6
100	+ 13.1	5,000	- 8.2
150	+ 10.3	6,000	- 9.6
200	+ 8.2	8,000	- 11.9
300	+ 5.5	10,000	- 13.7
400	+ 3.8	15,000	- 17.2
500	+ 2.6	20,000	- 19.6

Note: The RIAA EQ system operates in Constant Amplitude mode below the 500 Hz Turnover Frequency. It also operates in Constant Amplitude mode above the 2120 Hz Rolloff Frequency. The system operates in Constant Velocity mode between the Turnover and the Rolloff Frequencies.

Function Finder Table

In a program that has as many tools as both DC EIGHT and DC DC Forensics, we thought it might be helpful if you could look up the function to help you more easily find the tool you need.

Functions Unique to DC Forensics are shown in Italics

Function	Feature	Feature Location
3 Band EQ	Phono Pre Amp	Filter Menu
10 Band Graphic EQ	Graphic EQ	Filter Menu Under "EQ" under "Graphic EQ"
20 Band Graphic EQ	20 Band Graphic EQ	Filter Menu Under "EQ"
<i>30 Band Graphic EQ</i>	<i>30 Band Graphic EQ</i>	<i>Filter Menu Under "EQ"</i>
<i>32,000 Band Equalizer (FIR Based)</i>	<i>Spectral Filter</i>	<i>Forensics Menu</i>
33 RPM Record Click Filter	Expert or EZ Impulse Filter	Filter Menu

45 RPM Record Click Filter	Expert or EZ Impulse Filter	Filter Menu
78 RPM Record Click Filter	Expert or EZ Impulse Filter	Filter Menu
80 RPM Record Click Filter (Edison Diamond Disc or Pathé)	Expert or EZ Impulse Filter	Filter Menu
A-Law Compression	A-Law To .wav Conversion	Open/Save File As x.y
About DCart	About DCart	Help Menu
Acoustical Recording Transfer	Phono Pre Amp	Filter Menu
<i>Adaptive Filter</i>	<i>Adaptive</i>	<i>Forensics Menu</i>
<i>Adaptive Noise Rejection</i>	<i>Forensics AFDF</i>	<i>Continuous Noise Filter</i>
Add File to End of Existing File	Append To End	Edit Menu/Paste
ADPCM Compression	ADPCM to .wav Conversion	Open/Save File as x.y.
AIFF	Saving a file in the AIFF Format	File\Save As Menu
ALC (Multiband)	Punch & Crunch	Effects Menu
ALC (Wideband)	Dynamics Processor	Effects Menu
Ambience Enhancer	Reverb	Effects Menu
Ambience Reduction	CNF Spectral Subtraction	Continuous Noise Filter under Filter Menu
American 78 RPM Turnover Curve	Phono Pre Amp	Filter Menu
<i>Amplifier Non-linearity Reversal</i>	<i>Polynomial Filter</i>	<i>Forensics Menu</i>
Amplifier Non-linearity Simulation	Virtual Valve Amplifier	Effects Menu
Amplify Display	Right Slider Control	Next to Right Side of Display
Amplitude Measurement (Relative in dB)	VU Meter	View Menu
Annotate Printed Document	Page Setup	File Menu
Automatic Noise Reduction	EZ Clean Filter	Filter Menu
Automatically Assign Names To Tracks Using CD Internet Database	CD Data base (Internet)	File Menu
Averaging Filter	Averaging Filter	Filter Menu
Average Phase Angle Management	XY Display	View Menu
Band Pass (FIR Based)	Brick Wall Filter	Forensics Menu
Balance (audio sound level)	Virtual Phono Preamp	Filter Menu
Balance (gain controls)	File Conversion Filter	Filter Menu
Band Pass (IIR Based)	Band Pass Filter	Filter Menu
Band Pass Filter (Butterworth Response)	Band Pass Filter (Butterworth Checkbox)	Filter Menu
Band Pass Filter (Chebyshev	Band Pass Filter	Filter Menu

Response)	(Chebyshev Checkbox)	
Band Stop Filter (FIR Based)	Brick Wall Filter	Forensics Menu
Bass Control	Phono Pre Amp	Filter Menu
Bass Sound Enhancement	Fat Bass	VVA Under Effects Menu
Batch File Editor	Batch File Editor	Filter Menu
Batch Processing	Batch File Editor	Filter Menu
<i>Binaural DSS – Left Reference</i>	<i>Continuous Noise Filter</i>	<i>Filter Menu</i>
<i>Binaural DSS – Right Reference</i>	<i>Continuous Noise Filter</i>	<i>Filter Menu</i>
Both Channels Processed	L/R Icon	Toolbar
Break a File into Smaller Pieces at Markers	Chop File into Pieces	CD-Prep Menu
Brick Wall Filters	Brick Wall Filters	Forensics Menu
Broadcast .wav (BWF)	Create a File having the Broadcast .wav format	File\Save As Menu
Burn a CD ROM	Burn a CD	CD Prep Menu
Buzz Reduction	Harmonic Reject Filter	Filter Menu
Cascade Multiple Filters	Multi-Filter	Filter Menu
CD Creation	CD Burner	CD Prep Menu & DC Tune Library
<i>Cell Phone Noise Interference Reduction</i>	<i>Cell Phone Noise Filter</i>	<i>Forensics Menu</i>
<i>Cepstrum Plots</i>	<i>Voice ID</i>	<i>Forensics Menu</i>
Change Resolution of File	Change Sample Rate / Resolution	Edit Menu
Change Sampling Rate of File	Change Sample Rate / Resolution	Edit Menu
Channel Mixing / Blending	Channel Blender	Effects Menu
Channel Toolbar Activation	Channel Toolbar	View Menu
Chop File into Pieces at Markers	Chop File into Pieces	CD-Prep Menu
Clone a File	Create a replicate of the opened file	File Menu\Clone Source
Clipping Repair	De-Clipper	Forensics Menu
Columbia LP Curve (Early)	Phono Pre Amp	Filter Menu
Comb Filter 1	Multiple Notch Filters / Multi-Filter	Filter Menu
<i>Comb Filter 2</i>	<i>Spectral Filter</i>	<i>Forensic Menu</i>
Combine 2 Mono Files into a Stereo File	File Split and Re-Combine	Edit Menu
Compress File Size	Save as .mp3 or .wma or .ogg or .oga	File Menu
<i>Compressor (Instantaneous)</i>	<i>Polynomial Filter</i>	<i>Forensics Menu</i>
Compressor (multiband)	Punch & Crunch	Effects Menu
Compressor (wideband)	Dynamics Processor	Effects Menu
Context Sensitive Help	Point At Feature	Hit F1 Key
Convert a Stereo .wav file to Monophonic	File Conversions	Filter Menu

Convert CDs to MP3s	CD to MP3 Conversion	File Menu
Convert Monophonic .wav file to Stereo	File Converter	Filter Menu
Convert Random to Brown Noise	Multi-Filter (preset)	Filter Menu
Convert Random to Pink Noise	Multi-Filter (preset)	Filter Menu
Convert Random to Seismic Noise	Multi-Filter (preset)	Filter Menu
Convert Redbook Audio on a CD to a .wav file.	Rip CD Feature	File Menu
Convert Stereo .wav file to Stereo Reverse .wav file	File Converter	Filter Menu
Convert .wav file to AIFF Type	Save As	File Menu
Convert .wav file to MP3 Type	Save As	File Menu
Copy a Portion of a .wav file	Copy	Edit Menu
Corner Frequency vs. Time	Filter Sweeper	Effects Menu
Cracked Record Click Remover	Big Click Filter	Filter Menu
Create a Playlist	Open / Create Playlist	File Menu
Create Silence	Mute	Edit Menu
Create Your Own Filter	Multi-Filter	Filter Menu
Cross fade Two .wav files	Paste - Cross fade	Edit Menu
Data Base of Files (DC Tunes)	Open/Create Playlist	File Menu/View Menu
Data Disc Burner	Burn A Data Disc	File or CD Prep Menu
DC Offset Eliminator	High Pass Filter	Filter Menu/Preset
<i>DC Offset Generation</i>	<i>Polynomial Filter</i>	<i>Forensics Menu</i>
Deep Bass is Missing	Sub-harmonic Synthesizer	Effects Menu
De-Esser	Dynamics Processor	Effects Menu
Delay or Advance timing of One Channel compared to the other	Time Offset feature in File Conversions	Filter Menu
Delete .wav files	Delete Files	File Menu
Differentiator	High Pass Filter (Preset)	Filter Menu
<i>Direct X Plug-In Filters</i>	<i>Direct X Filters</i>	<i>Filter Menu</i>
<i>Disguise a Voice</i>	<i>Voice Garbler</i>	<i>Forensics Menu</i>
Display Colors	Preferences / Display	Edit Menu
Display Setup	Preferences / Display	Edit Menu
<i>Distortion Reduction</i>	<i>Polynomial Filter</i>	<i>Forensics Menu</i>
<i>Distortion Synthesizer</i>	<i>Polynomial Filter</i>	<i>Forensics Menu</i>
Dithering	Change Sample Rate / Resolution	Edit Menu
Draw Waveform Segment	Pencil Tool	Toolbar
<i>DSS – Delay Reference</i>	<i>Continuous Noise Filter</i>	<i>Filter Menu</i>
DTMF Filter	Paragraphic EQ	Filter Menu under “EQ”
Dynamic Enhancer	Dynamic Noise Filter	Filter Menu
Dynamic Noise Filter	Dynamic Noise Filter	Filter Menu
<i>Dynamic Spectral Subtraction (DSS)</i>	<i>Continuous Noise Filter</i>	<i>Filter Menu</i>

Dynamics Compressor (multiband)	Punch & Crunch	Effects Menu
Dynamics Compressor (wideband)	Dynamics Processor	Effects Menu
Dynamics Expander (multiband)	Punch & Crunch	Effects Menu
Dynamics Expander (wideband)	Dynamics Processor	Effects Menu
Echo 1	Reverb	Effects Menu
Echo 2	Time Offset Feature in File Converter	Filter Menu
Echo Chamber	Echo Effect	Effects Menu
Editing History	Fast-Edit History	View Menu
Enhance Audio Quality	EZ Enhancer	Effects Menu
Enhance Sibilance	Overtone Synthesizer	Effects Menu
European 78 Turnover Curve	Phono Pre Amp	Filter Menu
Evans Harmonic Reject	Harmonic Reject Filter	Filter Menu
Exciter (Harmonic)	Checkbox under the Virtual Valve Amplifier	Effects Menu
Exit the Program	Exit	File Menu or “X” on the Top Right of screen
<i>Expander (Instantaneous)</i>	<i>Polynomial Filter</i>	<i>Forensics Menu</i>
Expander (multiband)	Punch & Crunch	Effects Menu
Expander (wideband)	Dynamics Processor	Effects Menu
Export Presets	Manage Presets	Edit Menu
Extract Audio From AVI Video	Open Video Files	File Open
EZ Clean Filter	Automatic Impulse, Hiss and Hum Filters	Filter Menu
EZ Enhancer	Complex Audio Enhancement Effects	Effects Menu
EZ Forensics Filter	Adaptive Forensics Audio Filters	Forensics Menu
EZ Impulse Filter	Multiple & Adaptive Impulse Type Filters	Filter Menu
Fade In	Fade-In	Edit Menu
Fade Out	Fade-Out	Edit Menu
Fast Edit Mode	Preferences / General	Edit Menu
File Conversions	File Conversions Filter	Filter Menu
File Information	File Information	View Menu
File Time Reversal	Reverse File	Effects Menu
File Toolbar Activation	File Toolbar	View Menu
Filter Toolbar Activation	Filter Toolbar	View Menu
Find Peak Value	Spectrum Analyzer (Checkbox)	View Menu
<i>Formant Identification</i>	<i>Voice ID</i>	<i>Forensic Menu</i>
Frequency & Amplitude vs. Time	Spectrogram	Forensics Menu
Frequency Domain	Spectrum Analyzer	View Menu

Measurements		
<i>Frequency Doubler</i>	<i>Polynomial Filter</i>	<i>Forensics Menu</i>
Gain Change	Gain Change	Edit Menu
Gain Change vs. Time	Gain Change	Edit Menu
Gain Normalizer	Normalized Gain Scaling	CD-Prep
Gain vs. Time	Gain Change	Edit Menu
Generate Overtones (Evens)	Overtone Synthesizer	Effects Menu
Generate Overtones (Odds and Evens)	Virtual Valve Amplifier	
Generate Sub-harmonics	Sub-harmonic Synthesizer	Effects Menu
Harmonic Reject Filter	Harmonic Reject Filter	Filter Menu
Hear Removed Signal or Noise	Keep Residue Checkbox	In Filter Dialog Boxes where appropriate
High Pass (FIR Based)	Brick Wall Filter	Forensics Menu
High Pass (IIR Based)	High Pass Filter	Filter Menu
High Pass Corner Frequency vs. Time	Filter Sweeper	Effects Menu
High Pass Filter (Butterworth Response)	High Pass Filter (Butterworth Checkbox)	Filter Menu
High Pass Filter (Chebyshev Response)	High Pass Filter (Chebyshev Checkbox)	Filter Menu
High Resolution Frequency Response Contouring	30-Band Graphic EQ	EQ Menu
Highlight an Area of the .wav file	Left Mouse + Drag	Mouse
Hiss Reduction 1	Continuous Noise Filter	Filter Menu
Hiss Reduction 2	Dynamic Noise Filter	Filter Menu
Hiss Reduction 3	Hiss Filter	Hiss Filter in EZ Clean Filter
Hum Reduction	Notch Filter	Filter Menu
Import Presets	Manage Presets	Edit Menu
Impulse Filter (Easy)	EZ Impulse Filter	Filter Menu
Impulse Filter (Expert)	Expert Impulse Filter	Filter Menu
Insert a Segment into a .wav file	Paste - Insert	Edit Menu
Insert File at Beginning of Another File	Insert at Start	Edit Menu/Paste
Instant Audio Review	Flashback	Multi-Filter
Integrator	Low Pass Filter (Preset)	Filter Menu
Intermodulation Distortion Reduction	CNF used in Artifact Suppression Mode	Filter Menu
Interpolate a Portion of a .wav file	Paste - Interpolate	Edit Menu
Interpolate Both Channels	“I” Key on Keyboard	Paste Interpolate or Keyboard
Interpolate Left Channel Only	“J” Key on Keyboard	Keyboard

Interpolate Right Channel Only	“K” Key on Keyboard	Keyboard
Last 4 Files Opened	Listing Near the Bottom of the Menu	File Menu
Lateral Cut Record to Monophonic Conversion	File Conversions / Presets	Filter Menu
Lead Vocal Removal	Channel Blender	Effects Menu
Left Channel Process Only	L Icon	Toolbar
Limiter	Dynamics Processor Presets	Effects Menu
Log to Disc	Multi-Filter / Log to Disc Button	Filter Menu
Lossless File Compression	FLAC (.flac)	File Menu (Save As)
Loudness	Volume Control	View Menu
Loudness Maximizer	Punch & Crunch in Compressor Mode	Effects Menu
Low Pass (FIR Based)	Low Pass Filter	Forensics Menu
Low Pass (IIR Based)	Low Pass Filter	Filter Menu
Low Pass Corner Frequency vs. Time	Filter Sweeper	Effects Menu
Low Pass Filter (Butterworth Response)	Low Pass Filter (Butterworth Checkbox)	Filter Menu
Low Pass Filter (Chebyshev Response)	Low Pass Filter (Chebyshev Checkbox)	Filter Menu
Marker (Add)	Right Mouse Click on Display	Right Mouse Button
Marker (Annotate or Label)	Right Mouse Click on Label Marker	Right Mouse Button
Marker (Clear All)	Clear All Markers	Marker Menu
Marker (Clear an Individual Marker)	Right Mouse Click on Delete Marker	Right Mouse Button
Marker (Drop)	Drop a Marker	Marker Menu
Marker (Got Next One)	Got Next Marker	Marker Menu
Marker (Got Previous One)	Got Previous Marker	Marker Menu
Marker (Highlight in between two)	Double Left Mouse click between 2 markers	Left Mouse Button
Marker (Highlight in between two)	Highlight Marked Area	Marker Menu
Marker (Move)	Drag with Mouse	Left Mouse button
Markers (Lock all in Place)	Lock Markers	Markers Menu
Median Filter	Median	Filter Menu
Medium Resolution Frequency Response Contouring	20-Band Graphic EQ	EQ Menu
Mids Control	Phono Pre Amp	Filter Menu
Mix two .wav files together	Paste - Mix	Edit Menu
MME Drivers Setup	Preferences / Soundcard	Edit Menu
Monitor Input vs. Output	Bypass (Checkbox)	All Filter and Effects Dialog Boxes
Mono DSS – Delay Reference	Continuous Noise Filter	Filter Menu

Move .wav file from Destination to the Source Display	Make Destination the Source	File Menu
MP3 Encoder Setup	Preferences / MP3 Encoder	Edit Menu
Mu-Law Compression	Mu-Law To .wav conversion	Open/Save File As x.y
<i>Music from Speech Separator</i>	<i>DSS in the CNF</i>	<i>Filter Menu</i>
<i>Narrow Crackle Impulse Noise</i>	<i>Narrow Crackle Filter</i>	<i>Filter Menu</i>
<i>Narrowband Noise Rejection</i>	<i>30 Band EQ</i>	<i>EQ Menu</i>
<i>Noise Gate</i>	<i>Dynamics Processor Presets</i>	<i>Effects Menu</i>
Noise Reduction (Wideband)	Continuous Noise Filter	Filter Menu
<i>Non-linear Transfer Function</i>	<i>Polynomial Filter</i>	<i>Forensics Menu</i>
Normal Continuous Noise Filter	Continuous Noise Filter	Filter Menu
Normalize Loudness between Multiple .wav files	Auto Leveling Feature	Batch Processor
Notch (IIR Based)	Notch Filter	Filter Menu
Notch Filter vs. Time	Filter Sweeper	Effects Menu
Odds & Evens Harmonic Reject	Multi-Filter	Filter Menu
Odds Harmonic Reject	Harmonic Reject Filter	Filter Menu
Offset Display	Left Slider Control	Next to Right Side of Display
On Line Help	Help, Contents	Help Menu
Open a Playlist	Open / Create Playlist	File Menu
Open a .wav file into Destination Display	Open Destination	File Menu
Open a .wav file into Source Display	Open Source	File Menu
Overtone Synthesis (x 2)	Overtone Synthesizer	Effects Menu
Paragraphic EQ	Paragraphic EQ	Filter Menu under "EQ"
Parametric EQ	Paragraphic EQ	Filter Menu under "EQ"
Paste as a New .wav file	Paste as New File	Edit Menu
Paste Over a Portion of a .wav file	Paste - Over	Edit Menu
Paste Tone	Paste Bleep	Edit Menu/Paste
Pause Play	Play / Record Toolbar	Pause Button on Toolbar
Pencil Editing	Pencil Icon	Toolbar
Pencil Tool	Pencil Icon	Toolbar
Phase Inversion 1	File Conversions	Filter Menu
Phase Inversion 2	Channel Blender	Effects Menu
Phono Equalization Curves	Paragraphic EQ Presets	Filter Menu
Pitch Shift	Stretch & Squish Effect	Effects Menu
Place Markers Automatically on Silent spots	Find and Mark Silent Passages	CD-Prep Menu
Play a CD	DCTune Library	File Menu\DCTune Library

Play Controls Activation	Play Controls	View Menu
Play File	Play / Record Toolbar	Play Button / Spacebar
Play in a Loop	Loop Play	Loop Play Button on Toolbar
Playlists	DC Tune Library	File Menu/View Menu
Preview Filter or Effect	Preview Button	All Filters and Effects
Print Document	Print	File Menu
Print Preview	Print Preview	File Menu
Printer Setup	Print Setup	File Menu
Process in Batch Mode	Batch File Editor	Filter Menu
Quantize for CD Audio	Quantize for CD Audio	CD-Prep Menu
Ranking Filter	Median Filter	Filter Menu
Real Time Feed through	Multi-Filter / Live Preview Button	Filter Menu
Rebuild the Peak File (Waveform)	Rebuild Peak File	View Menu
Recorder	Record File	Edit Menu or Toolbar
<i>Rectifier</i>	<i>Polynomial Filter</i>	<i>Forensics Menu</i>
Reduce File Size	Save as .mp3 or .wma or .ogg or .oga	File Menu
Remove Portion of a .wav file	Cut	Edit Menu
Reverberation Simulation	Reverb	Effects Menu
Reverse RIAA EQ Curve	Paragraphic EQ Presets	Filter Menu
Review Real-time Audio	Flashback Mode	Multi-Filter
Rewind to Beginning	Rewind Button	Rewind Button on Toolbar
RIAA EQ Curve	Phono Pre Amp	Filter Menu
Right Channel Process Only	R Icon	Toolbar
Rip a CD	Rip CD Tracks	File Menu
Rumble Reduction 1	Continuous Noise Filter	Filter Menu
Rumble Reduction 2	High Pass Filter	Filter Menu
Run Function	Run Button	All Filters and Effects
Save a Destination Display .wav file	Save Destination As	File Menu
Scalar Amplitude Measurement	VU Meter	View Menu
Scratch & Crackle Filter	EZ Impulse	Filter Menu
Scrub Feature	Variable Speed Playback using the Mouse – Forward and Reverse	Toolbar
Selective Filtering	Sync Files	View Menu
Simulate Tubes and Transistors	VVA	Effects Menu
Sine Wave Synthesis	Make Waves	Edit Menu
Slot Filter (IIR Based)	Notch Filter	Filter Menu
Software Revision Number	About DCart	Help Menu
Sound Activated Recording	Record File / VOX Recording Checkbox	Edit Menu or Toolbar
Sound Card Set Up	Preferences	Edit Menu
Spectral Subtraction Filter	Continuous Noise Filter	Filter Menu

Spectral Inverse Filter	Normalize a Signal to Constant Power Spectral Density	Forensics Menu under Spectral Filter “EQ Mode” Selector
<i>Spectrogram</i>	<i>View Spectrogram</i>	<i>View Menu</i>
<i>Spectrograph Options</i>	<i>Preferences / Spectrograph</i>	<i>Edit Menu</i>
Spectrum Analyzer	Spectrum Analyzer	View Menu
Speech Filter (FIR based)	Brick Wall Filter (presets)	Forensics Menu
Speech Filter (IIR based)	Band Pass Filter (presets)	Filter Menu
Speed Correction	Change Speed Filter	Effects Menu
Splash Screen – On / Off	Preferences / Display	Edit Menu
Split Stereo File into 2 Mono Files	File Split and Re-Combine	Edit Menu
Square Wave Synthesis	Make Waves	Edit Menu
Static Remover	Expert Impulse Filter	Filter Menu
Status Bar Activation	Status Bar	View Menu
Stereo Simulation 1	File Converter (Time Offset)	Filter Menu
Stereo Simulation 2	Reverb	Effects Menu
Stereo Simulation 3	Paragraphic Equalizer, Presets	Filter Menu
Stroboscopes, Printable	Diamond Cut Install Folder	Strobe50Hz.wmf Strobe60Hz.wmf
Sub-Harmonic Synthesis (÷ 2)	Sub-harmonic Synthesizer	Effects Menu
Swept Waveform Synthesis	Make Waves	Edit Menu
Sync Files	Sync Files	View Menu
Synchronize Two .wav files	Sync Files	View Menu
Synthesize Round Bass	VVA	Effects Menu
Synthesize “Sweet” Treble	VVA	Effects Menu
Synthesize “Warm” Treble	VVA	Effects Menu
System Setup	Preferences	Edit Menu
System Status	Status Bar	View Menu (Bottom of Screen)
Tape Equalization Curves	Paragraphic EQ Presets	Filter Menu
THD Measurement	Spectrum Analyzer (Checkbox)	View Menu
Time at Cursor	Time Display	View Menu
Time Compensation Calculator and Corrector	Change Speed	Effects Menu
Time Compression	Stretch & Squish	Effects Menu
Time Delay	Echo Effect	Effects Menu
Time Expansion	Stretch & Squish	Effects Menu
Time Offset	File Conversions	Filter Menu
Time Span	Time Display	View Menu
Time Stamp the Segment	Multi-Filter / Checkbox	Filter Menu

Time Start	Time Display	View Menu
Time Stop	Time Display	View Menu
Timer Recording	Timer Record	Edit Menu
Timing Measurements	View Time Display	View Menu
Tip of the Day Activation	Tip of the Day / Checkbox	Help Menu
Tip of the Day De-Activation	Tip of the Day / Checkbox	Launch, and then after the Splash Screen
Tone Controls	Phono Pre Amp	Filter Menu
Top Octaves Missing	Overtone Synthesizer or Virtual Valve Amplifier	Effects Menu
Total Harmonic Distortion Measurement	Spectrum Analyzer (Checkbox)	View Menu
Treble Control	Phono Pre Amp	Filter Menu
Triangle Wave Synthesis	Make Waves	Edit Menu
Tube Simulator	Virtual Valve Amplifier	Effects Menu
Turnover Curves	Paragraphic EQ Presets	Filter Menu
Undo Edit	Undo	Edit Menu
Universal Impulse Filter	Expert Impulse Filter	Filter Menu
User Discussion Group (BBS / Forum)	User Discussion Group	Help Menu
User Preferences	Preferences	Edit Menu
Valve Simulator	Virtual Valve Amplifier	Effects Menu
Variable Frequency Response vs. Time	Filter Sweeper	Effects Menu
Variable Noise vs. Time	Adaptive Filter	Forensics Menu
Vector Measurement	XY Display	View Menu
Vertical Cut Record to Monophonic Conversion	File Conversions / Presets	Filter Menu
Vinyl LP Click Filter	Expert Impulse Filter	Filter Menu
Voice Activated Recording	Record File / VOX Recording Checkbox	Edit Menu or Toolbar
<i>Voice Print</i>	<i>Voice ID</i>	<i>Forensics Menu</i>
Volume Control	Volume Control	View Menu
Volume Control	Virtual Phono Preamp	Filter Menu
Vorbis (Conversion to Ogg Vorbis File format)	Save as .ogg or .oga	File Menu\Save As
<i>VOX Recording</i>	<i>Record File / VOX Recording Checkbox</i>	<i>Edit Menu or Toolbar</i>
VU Meter	VU Meter	View Menu
VU Meter Scale, Log or Linear	Preferences / General	View Menu
Weighted Median Filter	Median Filter	Filter Menu
White Noise Synthesis	Make Waves	Edit Menu
Wideband Noise Reduction	Continuous Noise Filter	Filter Menu
WMA Format File Saving	Save As .wma	File\Save As Menu
WMD Drivers Setup	Preferences / Soundcard	Edit Menu
XY Display	XY Display	View Menu
Zoom (Binary)	Zoom In or Out X2	View Menu or Toolbar

Zoom (Highlighted)	Zoom In or Out	View Menu or Toolbar
--------------------	----------------	----------------------

Hard Drive Space Recording Consumption

How much hard drive space will you need for your next recording? This handy chart should get you in the ballpark.

Digitization Disc Space Consumption as a function of Recording Mode and Sample Rate @ 16 bit resolution	
Sample Rate & Recording Mode	Mbytes per Minute
192 kHz Monophonic	22.500
192 kHz Stereophonic	45.000
96 kHz Monophonic	11.250
96 kHz Stereophonic	22.500
48 kHz Monophonic (Pro-Audio)	5.760
48 kHz Stereophonic (Pro-Audio)	11.520
44.1 kHz Monophonic	5.292
44.1 kHz Stereophonic (Compact Disc)	10.584
22.05 kHz Monophonic	2.646
22.05 kHz Stereophonic	5.292
16.000 kHz Monophonic (Forensics)	1.920
16.000 kHz Stereophonic (Forensics)	3.840
11.025 kHz Monophonic	1.323
11.025 kHz	2.646

Stereophonic	
--------------	--

Note 1 - Values are given for one process only (such as recording).

Note 2 - Values are given for 16-bit resolution only.

Note 3 - Multiply the above storage rates X 1.5 for 24-bit recording.

Note 4 - Windows currently operates with a 2 Gigabyte limit on .wav file size.

Hot Key Index

Here is a list of all known Hot Keys/Combos and their real world equivalent. These keyboard accelerators (sometimes just referred to as "Accelerators") are listed below:

"1"	Play selected area + 0.25 seconds on each side
"2"	Play selected area + 0.5 seconds on each side
"3"	Play selected area + 1 second on each side
"4"	Play selected area + 2 seconds on each side
"A"	Select displayed area (same as double click)
CONTROL+"A"	Select entire file
CONTROL+"C"	Copy selection to clipboard
"D"	Select destination
CONTROL+ "E"	Select/deselect the pencil (need to be zoomed in far enough to see waveform)
"I"	INTERPOLATE Full File (Bi-Modal {BM})
"J"	INTERPOLATE LEFT channel only {BM}
"SHIFT + I"	INTEROPLATE LEFT channel only {BM}
"K"	INTERPOLATE RIGHT channel only {BM}
CONTROL + "I"	INTERPOLATE RIGHTchannel only {BM}
"O"	INTERPOLATE (Time Domain Mode)
"L"	PLAY LOOPED
"M"	DROP MARKER
CONTROL+"M"	MUTE
"N"	GO TO NEXT MARKER
CONTROL+"N"	NEW File,
SHIFT+"N"	GO TO PREVIOUS MARKER
CONTROL+"O"	OPEN File
"P"	Open PREFERENCES
Plus Sign (+)	Zoom In X2
Minus Sign (-)	Zoom Out X2

CONTROL+"P"	PRINT
"Q"	Activates "Snap Selection to Zero Crossing"
CONTROL+"R"	Brings up Record screen
"S"	SELECT SOURCE
CONTROL+"S"	FILE SAVE
CONTROL+"U"	UNDO LAST EDIT
CONTROL+"V"	PASTE
ALT+BACKSPACE	UNDO
SHIFT+DELETE	CUT
END	REWIND TO Start
F1	Launch HELP file system
SHIFT+F1	CONTEXT SENSITIVE HELP
F6	NEXT PANE
SHIFT+F6	PREV_PANE
HOME	FORWARD TO END
CONTROL+INSERT	EDIT COPY,
SHIFT+INSERT	EDIT PASTE,
LEFT ARROW	NUDGE RIGHT edge of selected area to the left
RIGHT ARROW	NUDGE RIGHT edge of selected area to the right
SHIFT+LEFTARROW	NUDGE LEFT edge of sel. area to the left
SHIFT+RT ARROW	NUDGE LEFT edge of sel. area to the right
RETURN	Default button (this is the cancel button on all filters)
SPACEBAR	Play File or Record/Pause (toggle) when the record dialog box is active.
UP ARROW	Increments the selected parameters
DOWN ARROW	Decrements the selected parameters
CONTROL+"X"	CUT
"X"	ZOOM OUT
"Z"	ZOOM IN
CONTROL+"Z"	ZOOM OUT FULL
CONTROL+"B"	Paste Bleep
ALT+"S"	Toggles between Spectrogram& Normal Mode
Esc	Exits from Direct Spectral Edit (DSE) Mode or from the Spectrogram
CONTROL+"T"	Paste Insert

Human Hearing Frequency Response vs. Age

(0 dB is referenced to the 20 – 39 year old age group)
(Values are Averages for both Men and Women)

Age	<u>400 Hz</u>	<u>1 kHz</u>	<u>2 kHz</u>	<u>4 kHz</u>	<u>10 kHz</u>
<19	0 dB	0 dB	0 dB	+1 dB	+3 dB
20-29	0 dB	0 dB	0 dB	0 dB	0 dB
30-39	- 1 dB	- 2 dB	- 2 dB	- 3 dB	- 6 dB
40-49	-2 dB	- 3 dB	- 5 dB	-9 dB	-15 dB
50-59	- 4 dB	- 7 dB	- 13 dB	- 20 dB	- 30 dB
60-69	- 5 dB	- 12 dB	- 21 dB	- 32 dB	- 45 dB

Tape Speeds in Inches Per Second (ips)

The following is a listing of common speeds used by tape recorders:

Speed (IPS)	Pro Reel to Reel	Home Reel to Reel	Compact Cassette	Micro Cassette
30	X	-	-	-
15	X	-	-	-
7 1/2	-	X	-	-
3 3/4	-	X	-	-
1 7/8	-	-	X	X
15/16	-	-	-	X
15/32*	-	-	-	X

* This speed is also used by reel-to-reel analog data recorders.

Measurement Tools Table

Measurement	Stimulus System	Response System
Acoustical Signature	Calibrated microphone driving sound card input	Spectrograph
Aliasing Products	Swept Sine Wave (Make Waves Generator)	Spectrum Analyzer
Amplifier Linearity	Triangle Wave Generator	Waveform Display Window

	(Make Waves)	
Amplitude vs. Frequency	Any signal requiring analysis	Spectrum Analyzer
Amplitude vs. Time	Any signal requiring analysis	Waveform Display Window
Analog Tape Authentication	Look for line frequency or multiple line frequency spectral spikes or look for a higher order noise roll-off rate	Spectrum Analyzer operating in high resolution mode
Analog Tape Recorder Tape Head Azimuth Alignment	Azimuth Reference Tape	XY Display / Vector-scope
Ballistics Fingerprint	Audio Recording of ballistics events to be compared	Spectrograph
Average Phase Angle	Any Binaural Signal	Averaging Selection Box in the XY Display
Cross-talk	Sine Wave into One Channel from the Make Waves Generator; Terminate the opposite channels input with the proper impedance	Spectrum Analyzer or VU Meters
DC Offset	Terminate input with proper input impedance	Waveform Display Window
Dynamic Range	Any combination of signals requiring analysis	VU Meters with Peak Hold
Fall Time	Any signal requiring analysis	Waveform Display Window and Time Display with Markers
Frequency	Any signal requiring analysis	Spectrum Analyzer
Frequency & Amplitude vs. Time	Any signal requiring analysis	Spectrograph
Frequency Distribution	Any signal requiring analysis	Spectrum Analyzer
Frequency Ratio	Any signal requiring analysis	XY Display / Vector-scope
Frequency Response	Swept Sine Wave or Random Noise made by the Make Waves Generator	VU Meter (when using the Swept Sine Wave) or the Spectrum Analyzer (when using the Random Noise Generator)
Hard Disc Recording Time Available	Any signal being recorded	Recording Function
Instantaneous Frequency	Any signal requiring analysis	Waveform Display Window and Time Display with Markers. Calculate: $F = 1/t$
Intermodulation Distortion	Dual Sine Wave Tones made with the Make Waves Generator and summed together with "Paste Mix"	Spectrum Analyzer
Left Channel vs. Right Channel	Any signal(s) requiring analysis	XY Display / Vector scope
Linearity	Triangle Wave created using the Make Waves Generator	Waveform Display Window

Noise Floor	Properly Terminated Input	Spectrum Analyzer
Peak Amplitude	Any signal requiring analysis	VU Meter using the Peak Hold feature
Phase Angle	Any pair of signals having coherence	XY Display / Vector-scope
Phase Margin of equipment having a control loop system (1 st & 2 nd order)	Stereo Square Wave Generator (Make Waves)	Time display window (dampening factor of 'ring-out')
Power Amplifier Frequency Response vs. Output	Swept Sine Wave (Make Waves)	Proper Loading resistor and True RMS reading Voltmeter
Real Time of an Event	Any signal requiring analysis	Timer Recording with Time and Date Stamping
Recording Position	Any signal being recorded	Recording Function
Relative Amplitude (Scalar)	Any combination of signals requiring analysis	VU Meter
Relative Loudness	Any combination of signals requiring analysis	VU Meter
Rise Time	Any signal requiring analysis	Waveform Display Window and Time Display with Markers
Room Acoustical Balance	Random Noise Generator (Make Waves)	Spectrum Analyzer
Room Acoustical Propagation Delay & Reflection	Calibrated microphone driving a sound card input & an Impulse source like a handclap or an impulse created with the "Pencil" feature	Waveform Display Window, Markers and Time Display Feature
Signal to Noise Ratio	Sine Wave @ 0 dB vs. Noise Floor	Spectrum Analyzer
Slew Rate	Square Wave Generator (Make Waves)	Waveform Display Window and Time Display with Markers
Sound Card Performance	Please Refer to Application Note 2 (AN-2)	Please Refer to Application Note 2 (AN-2)
Sound Card Recording Level / Clipping	Audio signal applied to sound Card	Recording Function with recording VU meter and peak indicator
Stereo Separation	Sine Wave into One Channel from the Make Waves Generator; Terminate the opposite channels input with the proper impedance	Spectrum Analyzer, VU Meters or X-Y Display
Tape Recording Speed	Any analog tape recorded signal	Spectrum Analyzer
Time Derivative (dV / dt)	Any signal requiring analysis	Waveform Display Window and Time Display with Markers
Time Interval between Events	Markers	Time Display Feature

Total Harmonic Distortion (% THD)	Any Audio Device requiring performance testing using a Sine Wave stimulus (Make Waves)	Spectrum Analyzer with “Show THD” function and “Show Peak” enabled
Transient Response	Any Active Audio Device with a control loop with a Square Wave applied (Make Waves Generator)	Waveform Display Window
Turntable RPM	Neon or Fluorescent Lamp connected to a known Frequency Source	Printable Strobe discs found in the Help File
Voice Print comparison	Audio Signal of Voice(s) to be compared	Spectrograph
Wow and Flutter	Sine Wave (Make Waves)	Spectrum Analyzer

Rotary Head Tape Recorder Speeds:

- **DAT:**
0.321 ips (8.15 mm / sec)
- **VHS:**
1.31 ips - SP (Standard Play)
0.66 ips - LP (Long Play)
0.44 ips - EP (Extended Play)
- **Beta:**
1.58 ips (4.0 cm / sec) - Beta I
0.797 ips (2.0 cm / sec) - Beta II
0.524 ips (1.33 cm / sec) - Beta III
- **U-Matic:**
3.75 ips

Musical Scale

The following table provides the frequencies of four octaves of the tempered musical scale (1/2 step between notes) rounded in integers:

Note	Frequency	Note	Frequency
A	110	A (above middle C)	440
A# (B flat)	117	A# (B flat)	466
B	123	B	494
C (low C)	131	C (high C)	523
C# (D flat)	139	C# (D flat)	554
D	147	D	587
D# (E flat)	156	D# (E flat)	622
E	165	E	659
F	175	F	698
F# (G flat)	185	F# (G flat)	740
G	196	G	784
G# (A flat)	208	G# (A flat)	831
A (below middle C)	220	A (above high C)	880
A# (B flat)	233	A# (B flat)	932
B	247	B	988
C (middle C)	262	C	1,047
C# (D flat)	277	C# (D flat)	1,109
D	294	D	1,175
D# (E flat)	311	D# (E flat)	1,245
E	330	E	1,319
F	349	F	1,397
F# (G flat)	370	F# (G flat)	1,480
G	392	G	1,568
G# (A flat)	415	G# (A flat)	1,661
A (above middle C)	440	A	1,760

Note 1: Standard Pitch is based on the tone "A" of 440 Hz. With this standard, the frequency of Middle C should actually be 261.626 Hz.

Note 2: The entire Musical Scale from C0 (16.35 Hz) to D9# (9,956.06 Hz) are available as presets in the “Make Waves” Generator (Edit Menu).

Resistor Color Code

Standard RMA (Radio Manufacturers Association) Color Code:

Color	Significant Figure (Mantissa)	Decimal Multiplier (Exponent)
Silver	-	0.01
Gold	-	0.1
Black	0	1.0
Brown	1	10
Red	2	100
Orange	3	1,000 (1K)
Yellow	4	10,000 (10K)
Green	5	100,000 (100K)
Blue	6	1,000,000 (1M)
Violet	7	10,000,000 (10M)
Gray	8	100,000,000 (100M)
White	9	1,000,000,000 (1G)

Note 1: K = Kilo, M = Mega, G = Giga

Note 2: This color code scheme is sometimes used to identify the values of other electronic components and wires.

Sound Level

The following is a chart showing some sound sources and their Acoustic Power and Sound Power Levels measured at 10 meters from the source:

Sound Source (measured at 10 meters from source)	Total Acoustic Power (dB ref. to 10 ⁻¹² Watts) (A Weighted)	Power Level in dB (A Weighted)
Very soft Voice	1 NanoWatt	30
Conversational Voice	10 MicroWatts	70

Shouting Voice	1 MilliWatt	90
Auto on Highway	10 MilliWatts	100
Blaring Radio	100 MilliWatts	110
Piano	1 Watt	120
Small Aircraft Engine	3 Watts	125
Pipe Organ	100 Watts	140
75 Piece Orchestra	100 Watts	140
4 Propeller Airplane	1,000 Watts	150
Turbojet Engine	10,000 Watts	160
Ram-Jet Engine	100,000 Watts	170

Stroboscope

The following is a chart that you can use to create your own strobe disc using common line frequencies and RPM values:

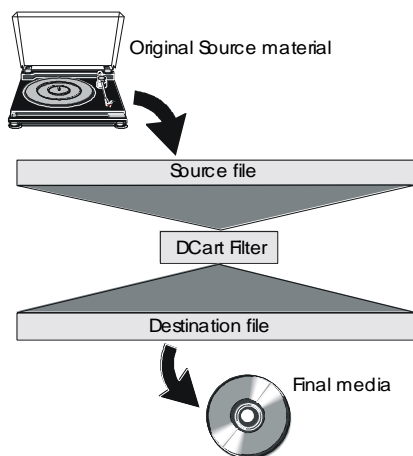
RPM	# of Divisions for 50 Hz	# of Divisions for 60 Hz
16	375	450
33.33	180	216
45	133	160
78.26	77	92
80	75	90

Note: Actually, two pulses of light are produced per cycle of the line by the power line. But, for improved visibility, it is better to use every other pulse to light up the strobe as is reflected by the chart above.

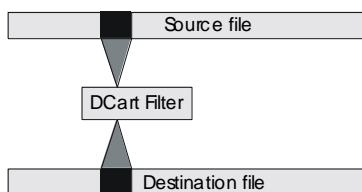
Note 2: Printable Stroboscope Disc Metafiles can be found in the install directory for your Diamond Cut software. The 50 Hz strobe disc is called “**Strobe50Hz.wmf**” and the 60 Hz version is called “**Strobe60Hz.wmf**”.

Sync Mode/Non Sync Mode Explanation Chart

The following diagram illustrates the standard filtering process of EIGHT/DC FORENSICS.



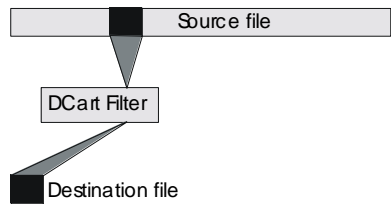
Sync Mode



Sync mode is the default mode of operation for the Classic Editing screen of *EIGHT/DC FORENSICS*. In Sync mode, both the Source and Destination files track each other. If you zoom into a section of the Source file, the Destination file will zoom to the same section. When you process the Source file using a *EIGHT/DC FORENSICS* filter, the program reads the Source file, processes it, and writes it to the

Destination file at exactly the same position as the Source file. This means that if you want to reprocess a section in the middle of a song, just highlight the section in the Source file that needs processing and run the filter again.

Non-Sync Mode



In Non-Sync mode, the highlighted section of the Source file is read and processed by the *EIGHT/DC FORENSICS* filter. The processed section is then written to the Destination file, starting at the beginning of the file. If a Destination file already exists, it will be overwritten (a prompt warns you of this). This mode is useful when only a section of the Source file needs to be extracted, or for testing a filter’s settings before processing an entire file.

Turnover Frequency Chart

Here is a listing of the most common turnover frequencies utilized by 78-RPM record brand and vintage:

Type, Brand, or Process	Turnover Frequency
Acoustical Recordings	0 Hz
Columbia (1925 - 1937)	200 Hz
Victor (1925 - 1937)	200 Hz
Westrex	200 Hz
Decca (1935 - 1949)	250 Hz
EMI	250 Hz
English Columbia	250 Hz
HMV (1931)	250 Hz
EMI (1931)	250 Hz

London	250 Hz
Blumlein	250 Hz
Columbia (1938 – End)	300 Hz
BSI	350 Hz
Capitol	400 Hz
Mercury	400 Hz
Brunswick	500 Hz
Decca (1925 – 1929)	500 Hz
Edison Laterals (1929)	500 Hz
MGM	500 Hz
Parlophone	500 Hz
Victor (1938 – 1952)	500 Hz
629	629 Hz

Note: Many of these Turnover Curves can be found as presets in the VPA or the Paragraphic EQ.

Wire Table

The Wire Table below is useful for calculating losses in power amplifier to speaker system cable connections.

(Standard Annealed Copper)

Wire Gauge in AWG	Wire Diameter(in Mils)	Resistance per Foot in Ohms (@ 20 degrees C)
0	324.9	0.00009827
1	289.3	0.0001239
2	257.6	0.0001563
3	229.4	0.0001970
4	204.3	0.0002485
5	181.9	0.0003133
6	162.0	0.0003951
7	144.3	0.0004982
8	128.5	0.0006282
9	114.4	0.0007921
10	101.9	0.0009989
11	90.74	0.001260
12	80.81	0.001588

13	71.96	0.002003
14	64.08	0.002525
15	57.07	0.003184
16	50.82	0.004016
17	45.26	0.005064
18	40.30	0.006385
19	35.89	0.008051
20	31.96	0.01015
21	28.46	0.01280
22	25.35	0.016140
23	22.57	0.02036
24	20.10	0.02567

Wire Gauge Rule of Thumb:

Conductor resistivity roughly doubles for every 3 AWG increase.
(Resistivity = 1 / Conductivity)

Note 1: The temperature coefficient of resistance for copper wire $\approx +0.41\%$ / degree C (4,100 ppm / degree C) referenced to 20 degrees C.

Note 2: The resistance of a 2 conductor cable will be need to be doubled to account for the round trip.

Note 3: Generally, audio power signals are carried with 18 AWG or lower except in some low power 25 or 70 Volt Constant Voltage audio distribution systems (from power amplifiers under 100 Watts).

Note 4: Except in some very unusual circumstances, no wire sizes higher than 16 AWG should be used to carry the output of an audio power amplifier to an audio system's loudspeakers, especially in permanent building installations.

Equalization Chart for LP Records

(Prior to the EQ Standardization in 1955)

<u>Label</u> <u>(Manufacturer)</u>	<u>Turnover</u> <u>Frequency in</u> <u>Hz</u>	<u>Roll-off at 10</u> <u>kHz in dB</u>
Angel	500	-13.7
Audio Fidelity	500	-16
Arizona	500	-13.7
Bach Guild	500	-16

Bartok	629	-16
Bethlehem	500	-13.7
Boston	500	-16
Caedmon	629	-16
Capitol	400	-12
Capitol-Cetra	400	-12
Cetra-Soria	500	-16
Classic Editions	500	-13.7
Clef	500	-13.7
Colosseum	400	-12
Columbia	300	-16
Concert Hall	400	-12
Decca	400	-12
Decca FFRR (1951)	300	-14
Decca FFRR (1953)	450	-11
Ducretet- Thompson	450	-11
EMS	375	-12
Epic	500	-16
Esoteric	400	-12
Folkways	500	-16
Haydn Society	500	-16
HMV	500	-16
Kapp	500	-13.7
London	450	-11
London International	450	-11
Lyrichord	500	-16
McIntosh	500	-13.7
Mercury	400	-12
MGM	500	-13.7
Montilla	500	-13.7
New Jazz	500	-13.7
Norgran	500	-13.7
Oceanic	500	-16
Oiseau-Lyre	500	-8.5
Overtone	500	-16

Polymusic	500	-16
Prestige	500	-13.7
RCA Victor (until 1953)	500	-13.7
Remington	500	-16
Riverside	500	-13.7
Romany	500	-13.7
Savoy	500	-13.7
Urania	500	-16
Vanguard	400	-12
Vox	400	-16
Westminster	400	-16

Note: Many of these curves can be found as presets within the VPA.

Language Translation (Deutsch/Español)

EIGHT/DC FORENSICS program terminology translations from English to German and Spanish

English	Deutsch	Español
About <i>Diamond Cut</i>	Über <i>DC-Art</i>	Acerca del <i>DC-Art</i>
Accuracy	Genauigkeit	Acuracia
Add Marker	Marker hinzufügen	Añadir marcador
Advanced Controls		Controles avanzados
Amplitude (dB)	Amplitude (dB)	Amplitud (dB)
Arrange Icons	Symbole anordnen	Arreglo de Icones
Attack (mSec)	Anstiegsflanke (mS)	Atacar (mSec)
Attenuation	Dämpfung	Atenuacion
Average	Mittelwert	Promedio
Averaging Filter	Mittelwertfilter	Determinacion del Promedio del Filtro
Band Pass	Bandpass	Banda de frecuencias alimentadas
Bandpass Filter	Bandpaßfilter	Filtro de Paso de Banda
Bandwidth	Bandbreite	Anchura de Banda
Bright	Hell	Brillante
Bypass	Bypass	Sobrepasar
Cancel	Abbrechen	Cancelar
Cascade	Fenster kaskadieren	Cascada
CD Preparation	CD Vorbereitung	Preparacion del CD
Center Frequency	Mittenfrequenz	Centro de Frecuencia
Change Speed	Geschwindigkeit ändern	Cambio de velocidad

Chop File into pieces	Datei aufteilen	Recortar el fichero en pedazos
Clear all Markers	Alle Marker löschen	Despejar todos los marcadores
Click Filter	Klickfilter	Golpecito seco en el Filtro
Clicks / Second	Klicks / Sekunde	Golpecitos en seco / Segundo
Close	Schließen	Fin
Close Destination	Zieldatei schließen	Cerrar Destinacion
Close Source	Quelldatei schließen	Cerrar las fuente de datos
Contents	Inhalt	Contenidos
Continuous Noise	Dauerrauschen	Sonido Continuo
Continuous Noise Filter	Rauschfilter	Filtrador Continuo de Sonido
Copy	Kopieren	Copia
Crossfade	Crossfade	Interseccion del aumento / disminucion del volumen alto
Curved	Kurve	Curvado
Cut	Ausschneiden	Cortar
Dark	Dunkel	Oscuro
DC-Art Progress	DC-Art Prozeß	Progreso del DC-Art
Decay	Ausklingzeit	Decaimiento
DeEsser	DeEsser	Normalizador de seseos
Delete	Löschen	Suprimir
Delete Files	Dateien löschen	Suprimir ficheros / archivos
Demo	Demo	Demostracion
Detail	Detail	Detalle
Device I/O Selection	Device I/O Auswahl	Seleccion del Dispositivo I / O
Drive	Drive	Impulsar
Dry	Trockenes Signal	Seco
Dynamic Noise	Dynamisches Rauschen	Sonido Dinamico
Dynamic Noise Filter	Dynamisches Rauschfilter	Filtro Dinamico de Sonido
Dynamics Processor	Dynamikprozessor	Procesor Dinamico
Early Level	Vorreflexion	Primera reverberacion de Amplificacion
Edit	Bearbeiten	Editar
Effects	Effekte	Efectos
Ending Pitch	Endtonhöhe	Terminando el Grado de Inclinacion
Even Only	Nur geradzahlige	Paridad Par
Exciter	Exciter	Excitador
Exit	Beenden	Salida
Expander/Gate	Expander/Gate	Ensanche / Puerta
Fade In	Einblenden	Aumento de Volumen
Fade Out	Ausblenden	Disminucion de Volumen
Fast Forward	Schneller Vorlauf	Avanzar / adelantar
File	Datei	Fichero / Archivo
File Conversions	Dateikonvertierungen	Traducciones del fichero

File Information	Dateiinformation	Informacion de Archivo
File Name	Dateiname	Nombre del fichero / archivo
File Toolbar	Datei Werkzeugleiste	Archivo de paneles
Files of Type	Dateityp	Modelos de ficheros
Filter	Filter	Filtro
Filter Freq.	Filterfrequenz	Filtro de frecuencia
Filter Harmonics	Oberwellen filtern	Filtro Armonicos
Filter Slope	Filterkurve	Filtro Inclinado
Filter Toolbar	Filter Werkzeugleiste	Filtro de instrumentos
Find and Mark Silent Passages	Stille Passagen finden und markieren	Encontrar y marcar los pasages silenciosos
Frequency	Frequenz	Frecuencia
Frequency Response	Frequenzgang	Frecuencia de respuesta
Frequency Spectrum	Frequenzspektrum	Frecuencia de espectros
From Mono to	Von Mono nach	Desde el Mono al
From Stereo to	Von Stereo nach	Desde el Stereo al
Full Range	Volle Bandbreite	Margen Entero
Gain	Verstärkung	Amplificacion
Gain Change	Verstärkungsänderung	Cambio de Volumen
Gain Normalize	Pegel normalisieren	Regularizar el sonido
Graphic Equalizer	Graphischer Equalizer	Equalizador Grafico
Harmonic Reject	Harmonische ausfiltern	Reyeccion Armonica
Harmonic Reject Filter	Oberwellen Unterdrückungsfilter	Filtro Armonico de Reyeccion
Help	Hilfe	Ayuda
High Frequency	hohe Frequenz	Frecuencia Alta
High Pass	Hochpass	Paso Alto
Highlight Marked Area	Markierten Bereich anwählen	Destacar las Areas Marcadas
Impulse Filter Parameters	Impulsfilter	Filtrar Parametros de Impulso
Impulse Noise	Impulsrauschen	Sonido de Impulso
Keep Residue	Rest behalten	Guardar residuo (resto)
Large	Groß	Grande
Left	Links	Mando izquierdo
Look In	Hineinsehen	Mirar En
Low Frequency	niedrige Frequenz	Frecuencia baja
Low Pass	Tiefpass	Avance de alimentacion de frecuencia baja
Lowpass Filter	Tiefpaßfilter	Filtro de pasaje bajo
Make Destination the Source	Ziel zur Quelle machen	Convertir los datos salidos del Fichero en datos nuevos de entrada de Fichero
Make Waves	Wellenformdarstellung	Hacer ondas
Marker	Marker	Marcador
Maximum Harmonic	Maximale Oberwelle	Armonia Maxima
Median	Zentralwert	Media
Median Filter	Mittelungsfilter	Filtro Medio

Medium	Mittel	Mediano
Mix	Mischung	Mixto
Mono	Mono	Mono
Mute	Stummschalten	Mudo
No	Nein	No
Noise Threshold	Rauschschwelle	Umbral de Sonido
Notch	Notchfilter	Hindumieno
Octave	Oktave	Octavo
Odd Only	Nur ungeradzahlige	Paridad impar
Open	Öffnen	Abrir
Open as Read Only	Schreibgeschützt öffnen	Abrir tal como se lee - Solo
Open Destination	Zieldatei öffnen	Salida abierta de archivo
Open Playlist	Spielliste öffnen	Lista de títulos
Open Source	Quelldatei öffnen	Abrir la fuente de datos
Operating Point (Quiescent Point)	Arbeitspunkt	Punto de Ininiazion
Optimization	Optimierung	Optimizacion
Output	Ausgang	Salida
Output Mix	Ausgangsmix	Mezcla de salidas
Overload	Übersteuerung	Sobrecarga
Paragraphic Equalizer	Parametrischer Equalizier	Equalizador Paragrafico
Paste	Einfügen	Pegar
Pause	Pause	Pausa
Pause File	Datei anhalten	Pausa del fichero / archivo
Pentode	Pentode	5 Elementos
Percent Done	Prozent erledigt	Porcentage acabado
Pitch Range	Tonhöhenbereich	Margen del Grado de Inclinacion
Play Controls	Transportkontrolle	Uso de los Controles
Play File	Datei spielen	Activar el fichero
Play Looped	Endlosschleife	Ejecutar el registro y repetir
Preferences	Voreinstellungen	Preferencias
Preview	Vorschau	Vostazo/Chequeo
Quantize for CD Audio	Für CD Audio quantisieren	Cuantificar para el audio CD
Ratio	Verhältnis	Proporcion
Record	Aufnahme	Registro
Record File	Datei aufnehmen	Registro del fichero / archivo
Reflections	Reflexionen	Reflejos
Release (mSec)	Abfallzeit (mS)	Librar (mSec)
Reset	Zurücksetzen (Reset)	Reajustar
Reset Levels	Pegel zurücksetzen	Reajustar los Niveles
Restoring a Recording	Aufnahme restaurieren	Restaurando el Registro
Restoring the Demo	Demo restaurieren	Restauracion del muestreo
Reverb	Hall	Reverberacion / Eco
Reverse	Rückwärts	Inverso

Reverse File	Datei rückwärts abspielen	Invertir fichero / archivo
Rewind	Zurückspulen	Rebobinar
Rewind File	Datei zurückspulen	Rebobinar el fichero / archivo
Right	Rechts	Mando derecho
Room Size	Raumgröße	Tamaño de espacio
Run Effect	Effekt abspielen	Comenzar el funcionamiento de Efecto
Run Filter	Filter anwenden	Comenzamiento del filtro
Save Source as	Gleiche Quelldatei wie	El mismo dato que el
Sample Noise	Quantisierungsrauschen	Ejemplo de Sonido
Samples	Samples	Ejemplos
Save	Speichern	Guardar
Save as Type	Speichern als	Guardar tal como se ha mecanografiado
Save Destination as	Zieldatei speichern als	Guardar destinacion como
Save In	Speichern in	Guardar En
Search for Help on	Suche Hilfe über	Busqueda de ayuda en
Select all	Alles auswählen	Seleccionar todo
Shape	Form	Form / Forma
Shift Threshold	Schwellwert verschieben	Cambio del Margen
Small	Klein	Pequeño
Source	Quelle	Fuente de Datos
Spectrum	Spektrum	Spektrum / Espectros
Spectrum Analyzer	Spektrumanalysator	Analizador de espectros
Speed	Geschwindigkeit	Velocidad
Starting Pitch	Anfangstonhöhe	Comienzo de Frecuencia
Status Bar	Zustandsleiste	Condicion de los paneles
Status Toolbar	Zustandsleiste	Compartimiento / Tablero de los paneles
Stereo	Stereo	Stereo
Stereo Reverse		Invertir el Stereo / retroceder atras Stereo
Stop File	Datei stoppen	Parar el fichero / archivo
Straight Line	Gerade	Linea Recta
Sweet	Süß	Agradable
Sync Files	Dateien synchronisieren	Sincronizacion de archivos
Threshold	Schwellwert	Margen
Tile	Fenster anordnen	Diseño de Azulejos
Time Offset	Zeitversatz	Cambio de tiempo
Tip of the Day	Tip des Tages	Consejo del día
Total Clicks Processed	Anzahl der bearbeiteten Klicks	Total de golpecitos secos (pinchazos/clicks) procesados
Total Samples to Process	Anzahl der zu bearbeiteten Samples	Total de ejemplos para procesar
Tracking	Gleichlauf	Rastreando

Transformer Coupled Class AB	Transformator gekopplete Class AB	Transformador Acoplado de Clase AB
Triode	Triode	Triodo
Tube Type/Configuration	Röhrentyp/Konfiguration	Clase de tubo / configuracion
Undo	Rückgängig	Deshacer
Very Large	Sehr groß	Muy Grande
View	Anzeigen	Vista
Vinyl LP	Vinyl LP	Vinilo LP
Virtual Valve Amplifier	Virtueller Röhrenverstärker	Amplificador de valvula virtual
Warm	Warm	Calido
Wet	Effektsignal	Mojado
Window	Fenster	Ventana
Yes	Ja	Si
Zoom to Markers	Zoom zwischen den Markern	Enforcar rapidamente
Zoom In	Zoom In	Enforcar
Zoom Out	Zoom Out	Desenfocar
# of Filters	Filteranzahl	Numero de filtro
% Change	% Veränderung	Porcentage de Cambio
% Selected Time	% Zeit	Porcentage del Tiempo seleccionado
2 Stage Class A	Zweistufige Class A	2 Procesos - Clase A
2 Stage Class AB	Zweistufige Class AB	2 Procesos - Clase AB

German Translation: Courtesy of Konstantin Themelidis

Spanish Translation: Courtesy of Monica Sanz Aznar (Monica Hash)

Preset Listings

10 Band Graphic EQ (see Graphic Equalizer – 10 Band)

20 Band Graphic EQ

- | | |
|---|--------------------------|
| 1. Alto Vocal Boost | 19. 4 kHz Slot Filter |
| 2. Alto Vocal Cut | 20. 40 Hz Slot Filter |
| 3. Baritone Vocal Boost | 21. 400 Hz Slot Filter |
| 4. Baritone Vocal Cut | 22. 5 kHz Slot Filter |
| 5. Bass Vocal Boost | 23. 50 Hz Slot Filter |
| 6. Bass Vocal Cut | 24. 500 Hz Slot Filter |
| 7. Bright Brass | 25. 6.2 kHz Slot Filter |
| 8. De-Boom | 26. 62 Hz Slot Filter |
| 9. Deep Bass | 27. 640 Hz Slot Filter |
| 10. Fat Bass | 28. 8 kHz Slot Filter |
| 11. Flat Response | 29. 80 Hz Slot Filter |
| 12. HF Boost with Shelf | 30. 800 Hz Slot Filter |
| 13. HF Cut with Shelf | 31. 1 kHz Notch Filter |
| 14. Less “Ess” | 32. 1.3 kHz Notch Filter |
| 15. Low Mid Range Bump-Up | 33. 1.6 kHz Notch Filter |
| 16. Low Mid Range Cut-Down | 34. 10 kHz Notch Filter |
| 17. Mid-band Bump | 35. 100 Hz Notch Filter |
| 18. Mid-band Dip | 36. 125 Hz Notch Filter |
| 19. Midrange Bump-Up | 37. 13 kHz Notch Filter |
| 20. Midrange Cut-Down | 38. 16 kHz Notch Filter |
| 21. Sad Face | 39. 160 Hz Notch Filter |
| 22. Smiley Face | 40. 2 kHz Notch Filter |
| 23. Soft Brass | 41. 2.5 kHz Notch Filter |
| 24. Soprano Vocal Boost | 42. 20 kHz Boost Filter |
| 25. Soprano Vocal Cut | 43. 200 Hz Notch Filter |
| 26. Sparkling Brilliance | 44. 25 Hz Notch Filter |
| 27. Tenor Vocal Boost | 45. 250 Hz Notch Filter |
| 28. Tenor Vocal Cut | 46. 31 Hz Notch Filter |
| 29. Ultrasonic Pierce | 47. 3.1 kHz Notch Filter |
| 30. Set All Sliders to Maximum Position | 48. 320 Hz Notch Filter |
| 31. Set All Sliders to Middle Position | 49. 4 kHz Notch Filter |
| 32. Set All Sliders to Minimum Position | 50. 40 Hz Notch Filter |

30 Band Graphic EQ

- | | |
|-------------------------|----------------------------------|
| 1. 1 kHz Slot Filter | 51. 400 Hz Notch Filter |
| 2. 1.3 kHz Slot Filter | 52. 5 kHz Notch Filter |
| 3. 1.6 kHz Slot Filter | 53. 50 Hz Notch Filter |
| 4. 10 kHz Slot Filter | 54. 500 Hz Notch Filter |
| 5. 100 Hz Slot Filter | 55. 6.2 kHz Notch Filter |
| 6. 125 Hz Slot Filter | 56. 62 Hz Notch Filter |
| 7. 13 kHz Slot Filter | 57. 640 Hz Notch Filter |
| 8. 16 kHz Slot Filter | 58. 8 kHz Notch Filter |
| 9. 160 Hz Slot Filter | 59. 80 Hz Notch Filter |
| 10. 2 kHz Slot Filter | 60. 800 Hz Notch Filter |
| 11. 2.5 kHz Slot Filter | 61. Alternating Phase 1 |
| 12. 20 kHz Boost Filter | 62. Alternating Phase 2 |
| 13. 200 Hz Slot Filter | 63. Alternating Phase Great 1 |
| 14. 25 Hz Slot Filter | 64. Alternating Phase Great 2 |
| 15. 250 Hz Slot Filter | 65. Alternating Phase Small 1 |
| 16. 31 Hz Slot Filter | 66. Alternating Phase Small 2 |
| 17. 3.1 kHz Slot Filter | 67. Anti Speech Filter 1 (Steep) |
| 18. 320 Hz Slot Filter | 68. Anti Speech Filter 2 |
| | 69. Bass Boost |
| | 70. Bass Boost with Shelf |
| | 71. Bass Boost 2 |

72. Bass Cut
73. Bass Cut 2
74. Bass Cut with Shelf
75. Brilliance (Treble Boost)
76. Flat Response
77. Low Bass Boost
78. Low Bass Cut
79. Midrange Bump
80. Midrange Dip
81. Negative Slope
82. Negative Slope with 2 Shelves
83. Randomized Pass Band
84. Rumble & Hiss Filter
85. Rumble Filter
86. Sad Face
87. Sad Face with 2 Shelves
88. Smiley Face
89. Smiley Face with 2 Shelves
90. Speech Filter 1
91. Speech Filter 2
92. Speech Filter 3
93. Speech Filter 4
94. Speech Filter 5
95. Speech Filter 6
96. Speech Filter 6 (Bottom & Top End Emphasis)
97. Treble Boost with Shelf
98. Treble Cut
99. Treble Cut with Shelf
100. Set All Sliders to Maximum Position
101. Set All Sliders to Middle Position
102. Set All Sliders to Minimum Position

Adaptive Filter

1. Basic Forensics Adaptive Filter
2. Basic Forensics Adaptive Filter with Unity Convergence
3. Basic Normalized Adaptive Filter
4. Binaural Filter with Left Channel Reference
5. Binaural Filter with Right Channel Reference
6. Sharp Adaptive Filter with delayed Monophonic Reference
7. Warbling Tone Rejection Filter

Adaptive Filter (Legacy Adaptive Filter)

1. Basic Forensics Adaptive Filter & - 5 dB Threshold
2. Basic Forensics Adaptive Filter & - 10 dB Threshold
3. Basic Forensics Adaptive Filter & - 15 dB Threshold
4. Basic Forensics Adaptive Filter & - 20 dB Threshold
5. Basic Forensics Adaptive Filter & - 20 dB Threshold
6. Basic Forensics Adaptive Filter & - 25 dB Threshold

7. Basic Forensics Adaptive Filter & - 30 dB Threshold
8. Basic Forensics Adaptive Filter & - 40 dB Threshold
9. Basic Forensics Adaptive Filter & - 50 dB Threshold
10. Basic Forensics Adaptive Filter & - 100 dB Threshold

Auto Voice Filter

1. Default
2. Heavy Attenuation
3. Maximum Attenuation
4. Minimum Attenuation
5. Moderate Attenuation
6. Nominal Attenuation

Averaging Filter

1. Heavy Handed Averaging Filter
2. Half Maximum Value Averaging Filter
3. Maximum Value Averaging Filter
4. Simple Cylinder Recording De-Hisser
5. 1950's Jukebox Sound
6. Light Surface Noise Remover

Band-pass Filter

1. 1930 Vintage Table Top Radio
2. 1950's Vintage Juke Box
3. AM Pocket Transistor Radio
4. Cardiac Sonograph Filter
5. Loud Walkman heard by person nearby
6. Megaphone
7. Metal Horn based Acoustical Phonograph
8. Modern "High-End" Audio System
9. Modern Cheap Table Top Radio
10. Night Club as heard in Parking Lot
11. Olde Acoustic Phonograph
12. Public Address System at Outdoor Event
13. Speech Filter
14. Stereo System heard in adjacent room
15. Telephone "off-the-hook" sound

Big Click Filter

1. Insensitive
2. Insensitive with De-Thumper
3. Most Insensitive
4. Most Insensitive with De-Thumper
5. Most Sensitive
6. Most Sensitive with De-Thumper
7. Nominal Sensitivity
8. Nominal Sensitivity with De-Thumper
9. Sensitive
10. Sensitive with De-Thumper

Brick Wall Filter

1. 1000 Hz Band-Stop Filter
2. 200 Hz High-pass Brick wall filter
3. 5000 Hz Low-pass Brick Wall Filter
4. 6000 Hz Low-pass Brick Wall Filter
5. 9000 Hz Low Pass Brick Wall Filter
6. Alpha Brainwave Bandpass Filter
7. Beta Brainwave Bandpass Filter

8. Theta Brainwave Bandpass Filter
9. Theta to Beta Brainwave Bandpass Filter
10. Basic Forensics Speech Filter
11. 60 Hz Band-stop Filter
12. Gentle Slope Forensics Speech Filter
13. Steep Slope Forensics Speech Filter
14. DTMF Band-pass Filter
15. Sub-Sonic Filter (Steep)

Cell Phone Noise Filter

1. Default
2. GSM – Aggressive Setting
3. GSM – Light Setting
4. GSM – Nominal Setting
5. GSM – Typical Starting Point
6. GSM – Very Aggressive Setting
7. GSM – Very Light Setting
8. Nextel – Aggressive
9. Nextel – Least Aggressive
10. Nextel – Typical Starting Point
11. Nextel – Very Aggressive

Change Speed Filter

1. Fractional Speed Mastering: 33.3 to 45 RPM
2. Fractional Speed Mastering: 45 to 78.2 RPM
3. Fractional Speed Mastering: 45 to 78.8 RPM
4. Fractional Speed Mastering: 45 to 80 RPM
5. Fractional Speed Mastering: 78.2 to 80 RPM
6. 0.5% Speed Decrease
7. 0.5% Speed Increase
8. 1.00 % Speed Decrease
9. 1.00 % Speed Increase
10. 1.50 % Speed Decrease
11. 1.50 % Speed Increase
12. 3.00 % Speed Decrease
13. 3.00 % Speed Increase
14. 6.00 % Speed Decrease
15. 6.00 % Speed Increase
16. 1.00 % Pitch Shift Upwards
17. 1.00 % Pitch Shift Downwards
18. Rise and Fall
19. Sin Wave Function
20. Sin Wave Function – Small change
21. Tape Recorder VOX Start Compensation
22. Tape Recorder VOX Stop Compensation

Channel Blender

1. Dreamy Veil
2. Early Stereo Anti “Ping-Pong” Effect
3. FM Stereo Multi-path Distortion Filter
4. FM Stereo Noise Filter
5. Lead Vocal Attenuator 1
6. Lead Vocal Attenuator 2
7. Lead Vocal Attenuator 3
8. Lead Vocal Attenuator 4
9. Severe FM Stereo Noise Filter
10. Stereo Ambience Signal Only
11. Vinyl LP Bass Clarifier

12. Vinyl LP Rumble Cancellation Filter

Continuous Noise Filter

1. Flat-line Response
2. CrO2 (High Bias) Cassette Tape Fingerprint (no NR)
3. CrO2 (High Bias) Cassette Tape Fingerprint (with NR)
4. Fe (Normal Bias) Cassette Tape Fingerprint (no NR)
5. FE (Normal Bias) Cassette Tape Fingerprint (with NR)
6. FeCRO2 (High Bias) Cassette Tape Fingerprint (no NR)
7. FeCRO2 (High Bias) Cassette Tape Fingerprint (with NR)
8. Metal (Metal Bias) Cassette Tape Fingerprint (no NR)
9. Metal (Metal Bias) Cassette Tape Fingerprint (with NR)
10. Demo Audio Wave file De-Noise
11. Dynamic Rumble (Only) Filter
12. Edison Diamond Disc Fingerprint
13. Lazy Man's Noise Sample
14. Pathé 80 RPM Acoustic Fingerprint
15. Typical 1940's Acetate Noise Fingerprint
16. Typical 1940's Shellac 78 Fingerprint
17. Typical 33.3 RPM Vinyl Noise Fingerprint
18. Typical 45 RPM Vinyl Noise Fingerprint
19. 1 7/8 ips reel-to-reel tape fingerprint
20. 3 3/4 ips reel-to-reel tape fingerprint
21. 3 3/4 ips reel-to-reel tape fingerprint (w/ enhancement)
22. 7 1/2 ips reel-to-reel tape fingerprint
23. 15 ips reel-to-reel tape fingerprint
24. 78 RPM, 8 inch Audiodisc Acetate Transcription
25. 78 RPM, 10 inch Audiodisc Acetate Transcription
26. 78 RPM, 10 inch Presto Acetate Transcription
27. 1890 Edison 2 Minute White Wax Cylinder
28. 1902 Columbia 78 RPM Fingerprint
29. 1903 Acoustical Lateral Fingerprint
30. 1904 Columbia Phonograph Record Fingerprint
31. 1904 Standard Disc Record 78 RPM Fingerprint
32. 1905 Victor 78 RPM Fingerprint
33. 1906 Edison 2 Minute Wax Cylinder (Standard)
34. 1912 Edison 2 Minute Celluloid Cylinder
35. 1919 Edison Blue Amberol 4 Minute Cylinder
36. 1928 Edison Lateral Cut 78 RPM Fingerprint
37. 1930's (Early) Victor 33 RPM Program Transcription Fingerprint
38. 1932 Victor Pre-Grooved Home Recording
39. 1933 HMV (His Masters Voice E) 78 RPM Fingerprint
40. 1939 English Columbia 78 RPM Fingerprint
41. 1940 Bluebird 78 RPM Fingerprint
42. 1940's Victor 78 RPM Fingerprint
43. 1948 Capitol 78 RPM Fingerprint
44. 1952 Vinyl 78 RPM Fingerprint
45. 1954 Monophonic 10 inch diameter LP Fingerprint
46. 1952 Mercury 78 RPM Fingerprint
47. 1973 Columbia Vinyl LP Fingerprint (RIAA Equalization)
48. Micro-cassette 1 7/8 ips Ferris Oxide Fingerprint
49. Micro-cassette 1 7/8 ips Metal Fingerprint
50. Micro-cassette 15/16 ips Ferris Oxide Fingerprint
51. Micro-cassette 15/16 ips Metal Fingerprint
52. Spectral Subtraction Sweet Spot
53. SW Radio Fingerprint – 16 Meter Band
54. Lazy Man's Noise Sample #2

56. 1938 Victor Scroll Label 78 Fingerprint
57. 1952 Vinyl 78 Fingerprint
58. 1954 Monophonic LP (10 Inch)
59. 1966 Roulette 45 RPM Fingerprint
60. 1971 RCA 45-RPM Fingerprint
61. 1973 Columbia LP (RIAA)
62. 1974 Columbia 45-RPM Fingerprint
63. Auto Spectrum CNF Aggressive Setting
64. Auto Spectrum CNF Maximum Setting
65. Auto Spectrum CNF Nominal Setting
66. Auto Spectrum CNF Very Aggressive Setting
67. Auto Spectrum CNF Very Light Setting

De-Clipper

1. Analog Tape Recorder De-Clipper
2. Digital Wave file De-Clipper
3. Operational Amplifier De-Clipper
4. Rate Occurrence De-Clipper
5. Severe Overdrive De-Clipper
6. Transistor Amplifier De-Clipper
7. Vacuum Tube Amplifier De-Clipper
8. Very Strong Interpolation Curvature

Dynamic Noise Filter

1. 3 ¼ ips Tape Hiss Attenuator
2. Brass Instrument Enhancer
3. Cassette Tape De-Hisser
4. Demo Audio Wave file De-Hiss
5. Forensics Wavefile Severe Noise Reduction Filter
6. Late 1930's 78-RPM Record Noise Reduction Filter
7. Spectral Exciter Enhancer 1
8. Spectral Exciter Enhancer 2
9. Spectral Exciter Enhancer 3
10. Spectral Exciter/Enhancer 1
11. Spectral Exciter/Enhancer 2

Dynamics Processor

1. 1 kHz De-Esser
2. 2 kHz De-Esser
3. 3 kHz De-Esser
4. 4 kHz De-Esser
5. 5 kHz De-Esser
6. 6 kHz De-Esser
7. 7 kHz De-Esser
8. 8 kHz De-Esser
9. 9 kHz De-Esser
10. 10 kHz De-Esser
11. ALC with Very Long Time Constants
12. Automatic Level Control
13. Automatic Level Control with Expander
14. Basic 2500 Hz De-Esser
15. ALC with Long Time Constants
16. ALC with Very Long Time Constants
17. Automatic Level Control
18. Automatic Level Control with Expander

19. Basic 2,500 Hz De-Esser
20. Elevator Music Compressor
21. Heavy Compression
22. Heavy Downward Expansion
23. Light Compression
24. Light Noise Reduction
25. Limiter
26. Noise Gate – Fast
27. Noise Gate – Slow
28. The Politicians Friend
29. ALC with Short Time Constants
30. Background Sound Enhancement
31. Basic 2,500 Hz De-Esser
32. Basic 2,500 Hz De-Esser
33. De-Ess #1
34. De-Ess #2
35. De-Ess #3
36. Downward Expander
37. General Purpose Compressor
38. Moderate Speed Ballad Compressor
39. Noise Gate
40. Noise Gate with 100 mSec Time Constants
41. Rock Compressor
42. Slow Ballad Compressor
43. Telephone "near-far" party compensation
44. The Sportscaster
45. The Wedding DJ

Echo Effect Presets

1. Asbury Park
2. Basketball Bounce
3. Esser 1
4. Esser 2
5. Forensics Speech Articulator 1
6. Forensics Speech Articulator 2
7. Forensics Speech Articulator 2
8. Grand Canyon
9. Harmony in 2 Parts
10. Harmony in 3 Parts
11. Little Canyon
12. Long and Narrow Club
13. New Orleans
14. Boise Buildup Effect
15. Phaser
16. Pole-Zero Pair
17. Reverse Echo
18. San Francisco Dramatic
19. San Francisco Very Light
20. Slapback
21. Small and Intimate Club
22. Small Club
23. Space Wars
24. Spring Reverb Light
25. Spring Reverb Loosely Wound
26. Spring Reverb Tightly Wound
27. St. Patrick's Cathedral
28. Stereo Heavy Plate Reverb
29. Stereo Light Plate Reverb

30. Stereo Ping-Pong Light
31. Stereo Ping-Pong Very Light
32. Stereo Reverse Light Plate Reverb
33. Stereo Reverse Ping Pong Light
34. Stereo Reverse Ping-Pong Very Light
35. Stereo Reverse Plate Reverb
36. Stereo Simulator 1
37. Stereo Simulator 2
38. Stereo Simulator 3
39. Stereo Simulator 4
40. Vocal Emphasize
41. Xenon

EZ-Clean Filter

1. EZ Clean (Aggressive)
2. EZ Clean (Average)
3. EZ Clean (Light Touch)
4. EZ Clean (Maximum)
5. EZ Clean (Minimum)
6. EZ Clean (Moderate – Light Hiss)
7. EZ Clean (Nominal Setting)
8. EZ Clean (Very Aggressive)
9. EZ Clean (Gentle Touch)
10. EZ Clean (Very Light Touch)
11. EZ Clean (Zero Effect)
12. 50 Hz Hum Filter Only
13. 60 Hz Hum Filter Only
14. Crackle Filter Only (Light)
15. Crackle Filter Only (Aggressive)
16. Crackle Filter Only (Nominal)
17. Crackle Filter Only (Very Aggressive)
18. Hiss Filter Only (Extremely Aggressive)
19. Hiss Filter Only (Gentle Touch)
20. Hiss Filter Only (Heavy Hand)
21. Hiss Filter Only (Light Touch)
22. Hiss Filter Only (Moderate Touch)
23. Hiss Filter Only (Very Heavy Hand)
24. Hiss Filter Only (Maximum)
25. Impulse Filter (Light)
26. Impulse Filter (Nominal)
27. Impulse Filter (Very Aggressive)
28. Impulse Filter (Aggressive)
29. Scratch Filter Only (Aggressive)
30. Scratch Filter Only (Light)
31. Scratch Filter Only (Nominal)
32. Scratch Filter Only (Very Aggressive)

EZ-Enhancer

1. Audiophile Quality
2. Automatic Level Control 1
3. Automatic Level Control 2
4. Automatic Level Control 3
5. Automatic Level Control 4
6. Default
7. Effects Bypass
8. Exciter – Evens Compressor
9. Exciter – Evens Expander
10. Exciter – Odds & Evens Compressor

11. Exciter – Odds & Evens Expander
12. Fat Bottom Evens, Balanced Compression
13. Fat Bottom Evens Balanced Expansion
14. Fat Bottom Evens, Compression
15. Fat Bottom Evens Expansion
16. Fat Bottom Evens Tubes
17. Fat Bottom Odds, Balanced Compression
18. Fat Bottom Odds Compression
19. Fat Bottom Odds Expansion
20. Fat Bottom Odds Tubes
21. Final Cleanup 33 RPM
22. Final Cleanup 45 RPM
23. Final Cleanup 78 RPM
24. Final Cleanup Tape
25. Full Spectrum Evens Tubes
26. Full Spectrum Odds Tubes
27. Multiband Limiter
28. Multiband Noise Gate
29. Slow Response Compressor
30. Slow Response Expander
31. Spoken Word Filter
32. Sweet Balanced Exciter
33. Sweet Evens Compression
34. Sweet Evens Exciter
35. Sweet Evens Expansion
36. Sweet Evens Tubes
37. Sweet Odds & Evens Exciter
38. Sweet Odds, Balanced Compression
39. Sweet Odds, Balanced Expansion
40. Sweet Odds Compression
41. Sweet Odds Expansion
42. Sweet Odds Tubes
43. Warm Balanced Exciter
44. Warm Evens, Balanced Compression
45. Warm Evens Balanced Expansion
46. Warm Evens Compression
47. Warm Evens Exciter
48. Warm Evens Expansion
49. Warm Evens Tubes
50. Warm Odds & Evens Exciter
51. Warm Odds, Balanced Compression
52. Warm Odds, Balanced Expansion
53. Warm Odds Compression
54. Warm Odds Expansion
55. Warm Odds Tubes

EZ-Impulse Noise Filter

1. 45-RPM Record Starting Point
2. 7 Inch, 33-RPM Starting Point
3. Aggressive Scratch & Crackle Remover
4. Aggressive Scratch and Crackle Remover
5. Crackle Only Eliminator
6. Gentle Scratch and Crackle Remover
7. Heavy Scratch, Light Crackle
8. Late 1940's 78
9. Light Scratch, Moderate Crackle Remover
10. Moderately Gentle Impulse Filter
11. Scratch and Crackle Eliminator

12. Scratch Only Eliminator
13. Very Aggressive Impulse Filter
14. Very Large 78 RPM Clicks
15. Very Large LP Clicks

EZ-Forensics Filter

1. (170 Filter Factory Presets which are available via a Wizard Driven System)

File Conversions

1. 1000 Hz, - 90 Degree Phase Shift Converter
2. 1000 Hz, + 90 Degree Phase Shift Converter
3. 400 Hz, +120 Degree Phase Shift Converter
4. 400 Hz, -120 Degree Phase Shift Converter
5. 50 Hz, + 90 Degree Phase Shift Converter
6. 50 Hz, +120 Degree Phase Shift Converter
7. 50 Hz, -120 Degree Phase Shift Converter
8. 50 Hz, -90 Degree Phase Shift Converter
9. 60 Hz, - 90 Degree Phase Shift Converter
10. 60 Hz, + 90 Degree Phase Shift Converter
11. 60 Hz, +120 Degree Phase Shift Converter
12. 60 Hz, -120 Degree Phase Shift Converter
13. Adaptive Demo
14. Mono to Stereo Simulator
15. Mono to Stereo Simulator 2
16. Mono to Stereo Simulator 3
17. Monophonic to Stereo
18. Monophonic Wave file Clone
19. Monophonic Wave file Clone w/ 3 dB Attenuation
20. Monophonic Wave file Clone with 3 dB Gain
21. Monophonic Wave file Clone with 6 dB Attenuation
22. Monophonic Wave file Clone with 6 dB Gain
23. Stereo Lateral Cut to Monophonic
24. Stereo Phase Inversion
25. Stereo Solo Vocal Reduction
26. Stereo to Monophonic
27. Stereo to Single Track Mono (left)
28. Stereo to Single Track Mono (right)
29. Stereo to Stereo Reverse
30. Stereo Vertical Cut to Monophonic
31. Stereo Wave file Clone
32. Stereo Wave file Clone w/ 3 dB Attenuation
33. Stereo Wave file Clone w/ 3 dB Gain
34. Stereo Wave file Clone w/ 3 dB Gain
35. Stereo Wave file Clone w/ 6 dB Attenuation
36. Stereo Wave file Clone with 6 dB Gain
37. Stereo Wave file Clone with 9.9 dB Gain
38. Strange Brew
39. Vertical Cut To Monophonic Converter

Filter Sweeper

1. 33.3 RPM Low Pass Masked Fade-In
2. 33.3 RPM Low Pass Masked Fade-Out

3. 78 RPM Low Pass Masked Fade-In
4. 78 RPM Low Pass Masked Fade-Out
5. Exponential High Pass Masked Fade-In
6. Exponential High Pass Masked Fade-Out
7. Exponential Low Pass Masked Fade-In
8. Exponential Low Pass Masked Fade-Out
9. Exponentially Swept Notch Filter
10. Linear High Pass Masked Fade-In
11. Linear High Pass Masked Fade-Out
12. Linear Low Pass Masked Fade-In
13. Linear Low Pass Masked Fade-Out
14. Linearly Swept Notch Filter
15. Log Low Pass Masked Fade-In
16. Log Low Pass Masked Fade -Out
17. Logarithmically Swept Notch Filter

Gain Change

1. 0.5 dB Gain Decrease
2. 0.5 dB Gain Increase
3. 1.0 dB Gain Decrease
4. 1.0 dB Gain Increase
5. 1.5 dB Gain Decrease
6. 1.5 dB Gain Increase
7. 10 dB Gain Decrease
8. 10 dB Gain Increase
9. 11 dB Gain Decrease
10. 11 dB Gain Increase
11. 12 dB Gain Decrease
12. 12 dB Gain Increase
13. 13 dB Gain Decrease
14. 13 dB Gain Increase
15. 14 dB Gain Decrease
16. 14 dB Gain Increase
17. 16 dB Gain Decrease
18. 16 dB Gain Increase
19. 17 dB Gain Decrease
20. 17 dB Gain Increase
21. 18 dB Gain Decrease
22. 18 dB Gain Increase
23. 19 dB Gain Decrease
24. 19 dB Gain Increase
25. 2 dB Gain Decrease
26. 2 dB Gain Increase
27. 20 dB Gain Decrease
28. 20 dB Gain Increase
29. 25 dB Gain Decrease
30. 3 dB Gain Decrease
31. 3 dB Gain Increase
32. 30 dB Gain Decrease
33. 4 dB Gain Decrease
34. 4 dB Gain Increase
35. 40 dB Gain Decrease
36. 5 dB Gain Decrease
37. 5 dB Gain Increase
38. 6 dB Gain Decrease
39. 6 dB Gain Increase
40. 60 dB Gain Decrease
41. 7 dB Gain Decrease

42. 7 dB Gain Increase
43. 78 RPM Laterals Gain Correction
44. 8 dB Gain Decrease
45. 8 dB Gain Increase
46. 80 dB Gain Decrease
47. 9 dB Gain Decrease
48. 9 dB Gain Increase
49. Curvilinear Fade-In
50. Curvilinear Fade-Out
51. De-Clipper Pre-Processing Gain Change
52. Fade Down / 20 dB
53. Fade Up / 20 dB
54. Logarithmic Fade In
55. Logarithmic Fade Out
56. Slew Gain Down from 0 dB to -10 dB
57. Slew Gain Down from 0 dB to -15 dB
58. Slew Gain Down from 0 dB to -20 dB
59. Slew Gain Down from 0 dB to -3 dB
60. Slew Gain Down from 0 dB to -4 dB
61. Slew Gain Down from 0 dB to -5 dB
62. Slew Gain Down from 0 dB to -6 dB
63. Slew Gain Down from 0 dB to -7 dB
64. Slew Gain Down from 0 dB to -8 dB
65. Slew Gain Down from 0 dB to -9 dB
66. Slew Gain Down from +10 dB to 0 dB
67. Slew Gain Down from +15 dB to 0 dB
68. Slew Gain Down from +20 dB to 0 dB
69. Slew Gain Down from +3 dB to 0 dB
70. Slew Gain Down from +4 dB to 0 dB
71. Slew Gain Down from +5 dB to 0 dB
72. Slew Gain Down from +6 dB to 0 dB
73. Slew Gain Down from +7 dB to 0 dB
74. Slew Gain Down from +8 dB to 0 dB
75. Slew Gain Down from +9 dB to 0 dB
76. Slew Gain Up from -10 dB to 0 dB
77. Slew Gain Up from -15 dB to 0 dB
78. Slew Gain Up from -20 dB to 0 dB
79. Slew Gain Up from -3 dB to 0 dB
80. Slew Gain Up from -4 dB to 0 dB
81. Slew Gain Up from -5 dB to 0 dB
82. Slew Gain Up from -6 dB to 0 dB
83. Slew Gain Up from -7 dB to 0 dB
84. Slew Gain Up from -8 dB to 0 dB
85. Slew Gain Up from -9 dB to 0 dB
86. Slew Gain Up from 0 dB to +10 dB
87. Slew Gain Up from 0 dB to +15 dB
88. Slew Gain Up from 0 dB to +20 dB
89. Slew Gain Up from 0 dB to +3 dB
90. Slew Gain Up from 0 dB to +4 dB
91. Slew Gain Up from 0 dB to +5 dB
92. Slew Gain Up from 0 dB to +6 dB
93. Slew Gain Up from 0 dB to +7 dB
94. Slew Gain Up from 0 dB to +8 dB
95. Slew Gain Up from 0 dB to +9 dB

Graphic Equalizer (10 Band)

1. 1000 Hertz Bump
2. 1000 Hertz Dip

3. 1000 Hertz Small Bump
4. 1000 Hertz Small Dip
5. 1000 Hertz Very Small Bump
6. 1000 Hertz Very Small Dip
7. 125 Hertz Bump
8. 125 Hertz Dip
9. 125 Hertz Small Bump
10. 125 Hertz Small Dip
11. 125 Hertz Very Small Bump
12. 125 Hertz Very Small Dip
13. 16000 Hertz Bump
14. 16000 Hertz Dip
15. 16000 Hertz Small Bump
16. 16000 Hertz Small Dip
17. 16000 Hertz Very Small Bump
18. 16000 Hertz Very Small Dip
19. 2000 Hertz Bump
20. 2000 Hertz Dip
21. 2000 Hertz Small Bump
22. 2000 Hertz Small Dip
23. 2000 Hertz Very Small Bump
24. 2000 Hertz Very Small Dip
25. 250 Hertz Bump
26. 250 Hertz Dip
27. 250 Hertz Small Bump
28. 250 Hertz Small Dip
29. 250 Hertz Very Small Bump
30. 250 Hertz Very Small Dip
31. 31 Hertz Bump
32. 31 Hertz Dip
33. 31 Hertz Small Bump
34. 31 Hertz Small Dip
35. 31 Hertz Very Small Bump
36. 31 Hertz Very Small Dip
37. 4000 Hertz Bump
38. 4000 Hertz Dip
39. 4000 Hertz Small Bump
40. 4000 Hertz Small Dip
41. 4000 Hertz Very Small Bump
42. 4000 Hertz Very Small Dip
43. 500 Hertz Bump
44. 500 Hertz Dip
45. 500 Hertz Small Bump
46. 500 Hertz Small Dip
47. 500 Hertz Very Small Bump
48. 500 Hertz Very Small Dip
49. 62 Hertz Bump
50. 62 Hertz Dip
51. 62 Hertz Small Bump
52. 62 Hertz Small Dip
53. 62 Hertz Very Small Bump
54. 62 Hertz Very Small Dip
55. 8000 Hertz Bump
56. 8000 Hertz Dip
57. 8000 Hertz Small Bump
58. 8000 Hertz Small Dip
59. 8000 Hertz Very Small Bump
60. 8000 Hertz Very Small Dip

61. Acoustical 78 EQ
62. Acoustical 78 EQ
63. Alternating Ripple
64. Alternating Ripple Reverse
65. Bass Boost
66. Bass Boost with Slight High End Cut
67. Bass Cut
68. Brilliance
69. Deep Bass
70. Flat
71. Front Row Center
72. High Definition
73. Loudness Contour
74. Mids Boost
75. Mids Cut
76. Noble
77. Presence
78. Robust Male Speaking Voice
79. Sad Face
80. Sharp Edge Remover
81. Smiley Face
82. Sub-Woofers Accentuation
83. Treble Boost
84. Treble Cut
85. Ultra Bass
86. Set All Sliders to Maximum Position
87. Set All Sliders to Middle Position
88. Set All Sliders to Minimum Position

Harmonic Reject Filter

1. 35 mm Cine Frame Audio Flicker Filter
2. American 120 Hz Buzz Filter
3. American 120 Hz Buzz Filter - Gentle touch
4. American 60 Hz Buzz Filter
5. American 60 Hz Buzz Filter - Gentle touch
6. European 100 Hz Buzz Filter
7. European 100 Hz Buzz Filter - Gentle touch
8. European 50 Hz Buzz Filter
9. European 50 Hz Buzz Filter - Gentle touch
10. Flange-1
11. Flange-2
12. Flange-3
13. Flange-4
14. Flange-5
15. NTSC TV Vertical Bleed-through Filter
16. PAL TV Vertical Bleed-through Filter
17. SECAM TV Vertical Bleed-through Filter

High-pass Filter

1. Demo Audio Wavefile De-Rumble
2. DC Offset Remover
3. De-Thumper (for selective filtering)
4. De-Thumper 2 (for selective filtering)
5. Differentiator
6. Impulse Viewer
7. Rumble Filter
8. Steep Rumble Filter
9. Steeper Rumble Filter

10. Sub-Sonic Filter
11. Sub-Sonic Filter 2
12. Sub-Sub Sonic Filter
13. Sub-Sub Sonic Filter 2

Impulse Noise Filter

1. 1940's Acetate Starting Point
2. 1940's Shellac 78 Starting Point
3. 45 RPM Starting Point
4. AM Radio Static Filter
5. Badly Gouged 78 De-Scratcher
6. Cylinder (2 minute) Record Starting Point
7. Cylinder (4 minute) Record Starting Point
8. Demo Audio Wave file De-Click
9. Diamond Disc 80 RPM Starting Point
10. Early Shellac 78 RPM Lateral Starting Point
11. High Fidelity 78s using HQ Mode
12. LP De-Click Starting Point
13. LP Static Discharge Noise Suppressor
14. Pathé 80 RPM Starting Point
15. Sharp Rise-time AM Radio Static Suppressor
16. Universal Impulse Filter
17. Vertical Cut Starting Point
18. Very Large 78-RPM Clicks
19. Very Light Touch 78 RPM De-Clicker
20. Vinyl 78 RPM Lateral Starting Point
21. Vinyl Extremely Small Click Only filter
22. Vinyl first pass using HQ mode
23. Vinyl Large and Dense Click Attenuator
24. Vinyl Large Click Only Filter
25. Vinyl Mode with Threshold Limit
26. Vinyl second pass using HQ mode
27. Vinyl Small Click Only Filter
28. Vinyl third pass using HQ mode
29. Vinyl Very Large Click Only filter
30. Vinyl Very Small Click Only Filter

Low-pass Filter

1. FM Multiplex Stereo Multi-path Filter
2. Integrator
3. Light LP Surface Noise Filter
4. Passive Analog Scratch Filter
5. Steep 78 RPM Surface Noise Attenuator
6. Steep LP Surface Noise Attenuator
7. Steep LP Surface Noise Attenuator
8. Ultrasonic Filter

Make Waves Generator

1. All Musical Notes running from C0 to D9#
2. 15 General Purpose Test Signals

Median Filter

1. 78 RPM Record De-Crackler
2. Extremely Muffled Recording Enhancer
3. Muffled Communications Enhancer
4. Timbre
5. Weighted Median Filter

Multi-Filter

1. 35 mm Cine Odd & Even Audio Flicker Filter
2. 45-RPM Record Cleanup Filter
3. A Room within a Room within a Room
4. Adaptive Filter with Noise Gate
5. Audio Mixer
6. Brick Wall Speech Enhancer with Gain Leveler
7. Brick Wall Speech Filter with Gain Leveler
8. Bright Tubes and Round Bass
9. Bright Tubes and Round Bass with Ambience
10. Butterworth Speech Filter w/Noise Gate
11. Butterworth Speech Filter With Gain Leveler
12. Butterworth Speech Filter with Gain Leveler
13. Cell Phone Noise Interference Attenuator (1 – 6)
14. De-Clipper (Mono)
15. De-Clipper (Stereo)
16. Dual Impulse Filter
17. Dual Impulse Filter 2
18. Dual Impulse Filter 3
19. Extreme 50 Hz Harmonic Reject Filter
20. Extreme 60 Hz Harmonic Reject Filter
21. EZ Forensics_Proto1
22. EZ Forensics_Proto2
23. Forensics Demo Cleanup Filter
24. Forensics Filter Lineup-Nominal
25. General Purpose Record Cleaner
26. General Purpose Vinyl De-Clicker
27. Intermodulation Distortion Test Filter, 262 Hz & 2 kHz
28. Intermodulation Distortion Test Filter, 60 Hz & 7 kHz
29. Live 50 Hz Buzz Filter
30. Live 50 Hz Buzz Filter (Odd and Even)
31. Live 60 Hz Buzz Filter
32. Live 60 Hz Buzz Filter (Odds & Evens)
33. Live Adaptive Filter
34. Live Average Filter
35. Live Band-pass Filter
36. Live Band-pass Filter
37. Live Brick Wall Filter
38. Live Channel Blender
39. Live Channel Blender
40. Live Channel Delay Line
41. Live Continuous Noise Filter
42. Live De-Clipper
43. Live Dynamic Noise Filter
44. Live Dynamics Processor
45. Live Echo
46. Live Ez-Impulse Noise Filter
47. Live File Converter
48. Live Graphic Equalizer
49. Live Hi-Pass Filter
50. Live Impulse Noise Filter
51. Live Low-Pass Filter
52. Live Median Filter
53. Live Mode Demo Clean-up SW Radio
54. Live Notch Filter
55. Live Paragraphic Equalizer
56. Live Polynomial Filter
57. Live Reverb
58. Live Spectral Filter
59. Live Tube Amplifier
60. Live Tube Amplifier
61. Loudness Maximizer
62. Low Cost Mono Cassette Tape Enhancer
63. Low Cost Mono Cassette Tape Enhancer
64. McGhee Filter
65. MP3 Enhancer / Full Tilt Boogie
66. MP3 Enhancer Four
67. MP3 Enhancer One
68. MP3 Enhancer Three
69. MP3 Enhancer Two
70. Multiple Notch Filter
71. Multiple Notch Filter
72. Olde Overloaded Radio Playing Far Away
73. Pink to White Noise Converter, 20 kHz (+/- 1.5 dB)
74. Poor Reception FM Stereo Enhancer
75. Record Clean-up Filter
76. Record Clean-Up Filter
77. Short Wave Radio Anti-Fader & Cleanup Filter
78. Speech Clarifier
79. Speech Enhancer (Basic)
80. Speech Filter with Noise Gate
81. Speech Filter with Noise Gate and Enhancer
82. Speech Filter with Noise Gate and Ultra-Enhancer
83. Stanton 500 RIAA Compensation Curve
84. Three Stage 50 Hz Buzz Filter
85. Three Stage 60 Hz Buzz Filter
86. Triple Impulse Filter
87. Ultra Flange
88. Universal Pop & Click Filter
89. Universal Pop and Click Filter
90. Very Bright Tubes
91. Very Bright Tubes
92. Very Scratch Vinyl De-Clicker
93. Victor Program Transcription Record (cleaner)
94. White (Random) to Brown Noise Converter
95. White Random To Seismic Noise Converter
96. White to 1/3 Octave Bucket Converter
97. White to Pink Noise Converter, 20 kHz
98. White to Seismic Noise Converter
99. White to Seismic Noise Converter, 100 Hz
100. White To Seismic Noise Converter, 20 Hz
101. White To Seismic Noise Converter, 50 Hz

Narrow Crackle Filter

1. Default
2. Gentle Touch Narrow Crackle
3. Large Sized Narrow Crackle
4. Medium Sized Narrow Crackle
5. Nominal Setting

6. Small Sized but Tall Narrow Crackle
7. Small Sized Narrow Crackle
8. Very Aggressive Narrow Crackle
9. Very Large Sized Narrow Crackle
10. Very Small Sized and Tall Narrow Crackle
11. Very Small Sized Narrow Crackle

Notch Filter

1. 1/3 Octave Bandpass Filter
2. 100 Hertz Slot Filter
3. 1000 Hz Slot Filter - One Tenth Octave
4. 1000 Hz Slot Filter- Half Octave
5. 1000 Hz Slot Filter- One Octave
6. 1000 Hz Slot Filter- Quarter Octave
7. 120 Hertz Slot Filter
8. 400 Hertz Notch Filter (Aircraft Electrical Noise Filter)
9. 400 Hertz Slot Filter
10. 50 Hz Slot Filter
11. 60 Hertz Slot Filter
12. American 120 Hertz Hum Filter
13. American 120 Hertz Hum Filter (sharp)
14. American 60 Hertz Hum Filter
15. American 60 Hertz Hum Filter (sharp)
16. American AM Heterodyne Filter
17. American AM Heterodyne Filter (sharp)
18. Endocardiograph Response Simulator
19. European 100 Hertz Hum Filter
20. European 100 Hertz Hum Filter (sharp)
21. European 50 Hertz Hum Filter
22. European 50 Hertz Hum Filter (sharp)
23. European AM Heterodyne Filter
24. European AM Heterodyne Filter (sharp)
25. FM Stereo Pilot Frequency Attenuator
26. Middle C Finder (262 Hertz on the Musical Scale)
27. NTSC Horizontal Scan Signal Attenuator (15,750 Hz)
28. PAL Horizontal Scan Signal Attenuator
29. SECAM Horizontal Scan Signal Attenuator
30. Simple 78 RPM de-crackle filter
31. Simple 78 RPM Record Hiss Filter

Overtone Synthesizer

1. Amplified Orchestral Effect
2. Balanced Orchestral Effect
3. Brilliant
4. Brilliant (very)
5. Default
6. Exaggerated Second Harmonic Overtones
7. Forensics Pseudo Sibilant (Clarifier)
8. Gentle Tough of Overtones
9. Near – Ultrasonic
10. Sparkling
11. Strong Second Harmonic Overtones
12. Subtle Second Harmonic Overtones
13. Top Octave Overtones Only
14. Top Octave Overtones Only (Strong)

15. Touch of Brilliance
16. Very Strong Second Harmonic Overtones
17. Wide Spectrum Second Harmonic Overtones
18. Wide Spectrum Second Harmonic Overtones (subtle)
19. Wide Spectrum Second Harmonic Overtones (very subtle)

Paragraphic Equalizer

1. 100 Hz Boost with 12 dB Shelf
2. 100 Hz Boost with 16 dB Shelf
3. 100 Hz Cut with -12 dB Shelf
4. 100 Hz Cut with -16 dB Shelf
5. 2 kHz Boost with 12 dB Shelf
6. 2 kHz Boost with 16 dB Shelf
7. 2 kHz Cut with -12 dB Shelf
8. 2 kHz Cut with -16 dB Shelf
9. 25 Hz Brick Wall High-pass Filter
10. 25 mSec De-Emphasis
11. 25 mSec Pre-Emphasis
12. 3 kHz Boost with 12 dB Shelf
13. 3 kHz Cut with -12 dB Shelf
14. 6 kHz Brick Wall Low-pass Filter
15. 75 mSec De-Emphasis
16. 75 mSec Pre-Emphasis
17. 78 RPM, 125 Hz Turnover Curve
18. 78 RPM, 200 Hz Turnover Curve
19. 78 RPM, 250 Hz Turnover Curve
20. 78 RPM, 500 Hz Turnover Curve
21. 78 RPM, 800 Hz Turnover Curve
22. 78-RPM, 300 Hz Turnover Curve
23. 78-RPM, 350 Hz Turnover Curve
24. 78-RPM, 400 Hz Turnover Curve
25. 78-RPM, 629 Hz Turnover Curve
26. American 120 Hz Hum Filter
27. American 120 Hz Hum Filter (sharp)
28. American 60 Hz Hum Filter
29. American 60 Hz Hum Filter (sharp)
30. Ampex Master EQ (AME) Playback Curve
31. CCIR 19cm / sec Home Playback Curve
32. CCIR 19cm / sec Home Recording Curve
33. CCIR 19cm/ sec Studio Playback Curve
34. CCIR 19cm/ sec Studio Recording Curve
35. CCIR 38cm/ sec Studio Playback Curve
36. CCIR 38cm/ sec Studio Recording Curve
37. DTMF Comb Filter (Narrow)
38. DTMF Comb Filter (Normal)
39. DTMF Comb Filter (Ultra-Wide)
40. European 100 Hz Hum Filter
41. European 100 Hz Hum Filter (sharp)
42. European 50 Hz Hum Filter
43. European 50 Hz Hum Filter (sharp)
44. Flat Response
45. Fletcher-Munson Aural Hot Spot Accentuator
46. Fletcher-Munson Aural Hot Spot Attenuator
47. Fletcher-Munson Contour @ 100 dB SPL
48. Fletcher-Munson Contour @ 120 dB SPL
49. Fletcher-Munson Contour @ 20 dB SPL

50. Fletcher-Munson Contour @ 40 dB SPL
51. Fletcher-Munson Contour @ 60 dB SPL
52. Fletcher-Munson Contour @ 80 dB SPL
53. Forensics Bar Room Compensation Filter
54. NAB Tape Playback Curve
55. Random IIR Based Phase Shifter 1
56. Random IIR Based Phase Shifter 2
57. Random IIR-Based Phase Shifter 1
58. Random IIR-Based Phase Shifter 2
59. Reverse NAB Tape Playback Curve
60. Reverse RIAA Phono Equalization Curve
61. Reverse RIAA w/ 125 Hz 78 Turnover
62. Reverse RIAA w/ 200 Hz 78 Turnover
63. Reverse RIAA w/ 250 Hz 78 Turnover
64. Reverse RIAA w/ 500 Hz 78 Turnover
65. Reverse RIAA w/ 800 Hz 78 Turnover
66. RIAA Phono Equalization Curve
67. Rolloff of 11 dB @ 10 kHz
68. Rolloff of 12 dB @ 10 kHz
69. Rolloff of 14 dB @ 10 kHz
70. Rolloff of 16 dB @ 10 kHz
71. Rolloff of 5 dB @ 10 kHz
72. Rolloff of 8.5 dB @ 10 kHz
73. Rumble Filter
74. Soft Slope (negative 1 dB / Octave)
75. Soft Slope (positive 1 dB / Octave)
76. Sound Level A-Weighting Curve
77. Sound Level C Weighting Curve
78. Stereo Simulator Left Channel Comb Filter
79. Stereo Simulator Left Channel Comb Filter (Wide)
80. Stereo Simulator Right Channel Comb Filter
81. Stereo Simulator Right Channel Comb Filter (Wide)

Reverb

1. Large Auditorium
2. Large Cathedral
3. Large Cavern
4. Large Submarine
5. Large Train Station
6. Light Reverb
7. Lighthouse
8. Lots of Reverb
9. Medium Size Auditorium
10. Medium Size Wood Room 1
11. Medium Size Wood Room 2
12. Phasing Effect 1
13. Phasing Effect 2
14. Slap-back 1
15. Slap-back 2
16. Slap-back 3
17. Slap-back 4
18. Small Intimate Night Club 1
19. Small Intimate Night Club 2
20. Small Intimate Night Club 3
21. Small Masonry Room 1
22. Small Masonry Room 2

23. Small Stone Church
24. Small Theater 1
25. Small Theater 2
26. Small Wood Church
27. Small Wood Room
28. Ultimate Reverb
29. Big Brass Boomy Bass
30. Bright Boomy Bass
31. Grande Canyon
32. Saint Peters Basilica
33. Boomy Olde Wood Room
34. Delay Line, Very Short
35. Delay Line, Short
36. Delay Line, Medium
37. Delay Line, Long
38. Echo 1
39. Echo 2
40. Echo 3
41. Echo 4
42. Ambience 1
43. Ambience 2
44. Ambience 3
45. Ambience 4
46. Gentle Touch 1
47. Gentle Touch 2
48. Metal Plate Reverb
49. Classical Concert Hall
50. Smokey Club

Polynomial Filter

1. 5th Order Asymmetrical Compressing Non-Linearity
2. 5th Order Asymmetrical Expander Non-Linearity
3. Asymmetrical Expanding Non-Linearity with Phase Inv.
4. Basic Asymmetrical Non-Linearity
5. Class A Amplifier
6. Comb Generator
7. Compress +, Expand -
8. Double Inflection Point Non-Linearity
9. Double Inflection Point Non-Linearity #2
10. Even + Odd Order Harmonic Generator
11. Even Order Frequency Multiplier
12. Even Order Frequency Multiplier 2
13. Four Inflection Points #1
14. Four Inflection Points #2
15. Four Inflection Points #2
16. Full Wave Rectifier
17. Gross Non-Linearity
18. Instantaneous Compressor
19. Instantaneous Expander
20. Inverse Class A Amplifier
21. Inverse Push-Pull Power Amplifier
22. Inverse Push-Pull Power Amplifier with Phase Inv.
23. Inverting Real Time Expander
24. Level Sensitive Fuzz Box
25. Negative DC Offset
26. Phase Inverter
27. Positive DC Offset
28. Real Time Compressor 2

29. Real Time Expander 2
30. Saturated Amplifier with Local Feedback
31. Sensitive Full Wave Rectifier
32. Sensitive Half Wave Rectifier
33. Slightly Expansive Non-Linearity
34. Very Expansive Non-Linearity
35. Very Kinky
36. Weak Bulbs in the Olde Radio Set
37. X 0.25 Gain Increase
38. X 0.25 Gain Increase With Expansion
39. X 4 Gain Increase

Punch and Crunch

1. 1950's Jukebox
2. AM Radio De-Compressor
3. Background Music Compressor
4. Big Bad Bass
5. Big Brash Bass
6. Big Brass Bass
7. Cheap Cassette Tape Recorder Dynamic Expander
8. Classical Music Compressor
9. Exciter
10. Fat Round Bottom
11. Female Lead Vocal Enhancer
12. Fission Reaction
13. FM Radio Dial Presence
14. Forensics (Noise Reduction)
15. Front Row, Center
16. Fusion Reaction
17. Heavy Compression / Balanced Spectrum
18. Heavy Expansion / Balanced Spectrum
19. Heavy Expansion / Bright
20. Heavy Expansion / Brilliant
21. Heavy Expansion / Round Bottom
22. Heavy Expansion Big Bass
23. Hip Hop
24. Light Compression / Balanced Spectrum
25. Light Expansion / Balanced Spectrum
26. Light Expansion / Bright
27. Light Expansion / Brilliant
28. Light Expansion / Round Bottom
29. Male Lead Vocal Enhancer
30. Moderate Compression / Balanced Spectrum
31. Moderate Compression / Muted
32. Moderate Expansion / Balanced Spectrum
33. Moderate Expansion / Bright
34. Moderate Expansion / Brilliant
35. Moderate Expansion / Round Bottom
36. Multi-band ALC
37. Very Heavy Compression / Balanced Spectrum
38. Very Heavy Compression / Muted
39. Very Heavy Expansion / Bright
40. Very Heavy Expansion / Brilliant
41. Very Light Compression / Balanced Spectrum
42. Very Light Compression / Muted

43. Very Light Expansion / Balanced Spectrum
44. Very Light Expansion / Brilliant

Spectral Filter

1. 100 Hz Hum Filter (for Defective European Filter Capacitors)
2. 120 Hz Hum Filter (for Defective US Filter Capacitors)
3. 400 Hz Aircraft Electrical Noise Filter
4. 50 Hz Hum Filter (European)
5. 60 Hz Hum Filter (US)
6. 60 Hz Buzz Filter
7. 5 kHz Brick Wall Low Pass Filter
8. Speech Filter
9. Speech and 400 Hz Aircraft Electrical Buzz Filter
10. Speech and 50 Hz Buzz Filter
11. Speech and 60 Hz Buzz Filter
12. Phase Shifter- Randomized
13. Phase Shifter
14. 400 Hz Aircraft Electrical Buzz Filter
15. Human Voice Filter
16. Male Voice Contour
17. Female Voice Contour
18. Auto EQ Forensic Normalize to Brown Noise
19. Auto EQ Forensic Normalize to Inverse Brown Noise
20. Auto EQ Forensic Normalize to Human Voice
21. Auto EQ Forensic Normalize to Inverse Human Voice
22. Auto EQ Forensic Normalize to Pink Noise
23. Auto EQ Forensic Normalize to Inverse Pink Noise
24. Auto EQ Forensic Normalize to Pink Noise, Full
25. Auto EQ Forensic Normalize to Inverse Pink Noise
26. Auto EQ Forensic Normalize to White Noise
27. Auto EQ Forensic Normalize to White Noise, Brick Wall
28. Auto EQ Forensic Normalize to Brown Noise, Full
29. Auto EQ Forensic Normalize to White Noise, Narrow

Stretch & Squish Filter

1. 10 % Faster
2. 10 % Slower
3. 15% Faster
4. 15% Slower
5. 2% Faster
6. 2% Slower
7. 20 % Faster
8. 20 % Slower
9. 30 % Faster
10. 30 % Slower
11. 5 % Faster
12. 5 % Slower
13. Constant Pitch
14. Disguised Voice Effect 1
15. Disguised Voice Effect 2
16. Disguised Voice Effect 3
17. Disguised Voice Effect 4
18. Disguised Voice Effect 5
19. Disguised Voice Effect 6
20. Higher Pitch and Speed
21. Lower Pitch and Speed
22. Much Higher Pitch and Speed
23. Much Lower Pitch and Speed

24. Spoken Word Transcription Rate
25. Spoken Word Transcription Rate #3
26. Spoken Word Transcription Rate 1
27. Spoken Word Transcription Rate 2

Sub-harmonic Synthesizer

1. Balanced Spectrum, Bump and Thump
2. Balanced Spectrum, Maximus Bump and Thump
3. Balanced Spectrum, Minimus
4. Balanced Spectrum, Nominal Level
5. Balanced Spectrum, Slight Bump and Thump
6. Deep Bass Balanced
7. Deep Bass Light
8. Deep Bass Maximus Thumpus
9. Deep Bass Minimus
10. Deep Bass Moderato
11. Default
12. Disco Bass
13. Terra Bass Balanced Level
14. Terra Bass Dance Club Atmosphere
15. Terra Bass Maximus Thumpus
16. Terra Bass Minimus

Virtual Phono Preamp (VPA)

1. Approximate AES LP EQ Curve
2. Approximate FFRR LP EQ Curve
3. Approximate NAB LP EQ Curve
4. Approximate Atlantic LP Label
5. Approximate Bartok LP Label
6. Approximate Blue Note Jazz LP Label
7. Approximate Canyon LP Label
8. Approximate Capitol – Cetra LP Label
9. Approximate Capitol LP Label
10. Approximate Concert Hall LP Label
11. Approximate Contemporary LP Label
12. Approximate Cook LP Label
13. Approximate Elektra LP Label
14. Approximate EMS LP Label
15. Approximate Folkways LP Label
16. Approximate Good – Time Jazz LP Label
17. Approximate L’Oiseau-Lyre LP Label
18. Approximate London LP Label
19. Approximate Lyricord LP Label
20. Approximate Mercury LP Label
21. Approximate Oceanic LP Label
22. Approximate Philharmonic LP Label
23. Approximate Polymusic LP Label
24. Approximate RCA Victor LP Label (Early)
25. Approximate Remington LP Label
26. Approximate Urania LP Label
27. Approximate Westminster LP Label
28. Exact Allied LP Label
29. Exact American Record Society LP Label
30. Exact Angel LP Label
31. Exact Bach Guild LP Label
32. Exact Bethlehem Label
33. Exact Boston LP Label

34. Exact Caedmon LP Label
35. Exact Camden LP Label
36. Exact Cetra – Soria LP Label
37. Exact Classic Editions Label
38. Exact Clef Label
39. Exact Collosseum LP Label
40. Exact Columbia LP Label
41. Exact Decca LP Label
42. Exact Epic LP Label
43. Exact Esoteric LP Label
44. Exact Haydn Society LP Label
45. Exact Kapp Label
46. Exact McIntosh Label
47. Exact MGM Label
48. Exact Montilla Label
49. Exact New Jazz Label
50. Exact Pacific Jazz LP Label
51. Exact Prestige Label
52. Exact RCA Victor Label
53. Exact Riverside LP Label
54. Exact Romany LP Label
55. Exact Savoy LP Label
56. Exact Tempo LP Label
57. Exact Vanguard LP Label
58. Exact Vox LP Label
59. Exact Walden LP Label
60. Default
61. Flat Preamp Hardware playing Acoustical Records
62. Flat Preamp Hardware playing American 78s
63. Flat Preamp Hardware playing Columbia Vinyl LPs
64. Flat Preamp Hardware playing European 78s
65. Flat Preamp Hardware playing RIAA Vinyl LPs
66. Line Input – Phono Preamp Bypass
67. RIAA Preamp Hardware Playing Acoustical Records
68. RIAA Preamp Hardware Playing American 78s
69. RIAA Preamp Hardware Playing Columbia Vinyl LPs
70. RIAA Preamp Hardware Playing European 78s
70. RIAA Preamp Hardware Playing RIAA Vinyl LPs
72. Approximate RIAA EQ with DIN Comp (Flat Preamp)
73. Approximate RIAA EQ with DIN Comp (RIAA Preamp)

Virtual Valve Amplifier (VVA)

1. 12AT7 based Class A Amplifier, Full Range
2. 12AU7 Based Class A Amplifier (Full Range)
3. 12AU7 Based Class A Amplifier (Sweet)
4. 12AU7 Based Class A Amplifier (Warm)
5. 12AU7 based Class A Amplifier, Full Range
6. 12AX7 Based Class A Amplifier (Full Range)

7. 12AX7 Based Class A Amplifier (Sweet)
8. 12AX7 Based Class A Amplifier (Warm)
9. 1935 Retro 15 Watt Push-Pull Amp
10. 1938 Retro 4 Watt Class A Power Amp
11. 25 Watt / Channel Power-Amp 1 (6L6-GC)
12. 25 Watt / Channel Power-Amp 2 (6L6-GC)
13. 25 Watt / Channel Power-Amp 3 (6L6-GC)
14. 25 Watt / Channel Power-Amp 4 (6L6-GC)
15. 2A3 Based Class A Amplifier (Full Range)
16. 2A3 Based Class A Amplifier (Sweet)
17. 2A3 Based Class A Amplifier (Warm)
18. 2A3 based Class A Amplifier Full Range
19. 6 EJ7B based Class A Amplifier, Full range
20. 6EJ7 Based Class A Amplifier (Full Range)
21. 6EJ7 Based Class A Amplifier (Sweet)
22. 6EJ7 Based Class A Amplifier (Warm)
23. 8 Watt, 6L6 Single-Ended Amplifier 1
24. 8 Watt, 6L6 Single-Ended Amplifier 2
25. 8 Watt, 6L6 Single-Ended Amplifier 3
26. Bright & Brassy Brass
27. Grunge (Odds Only)
28. Harmonic Enhancer 1 (Evens + Odds)
29. Harmonic Enhancer 2 (Evens + Odds)
30. Harmonic Enhancer 3 (Evens Only)
31. Harmonic Enhancer 4 (Evens Only)
32. Harmonic Enhancer 5 (Evens Only)
33. Harmonic Enhancer 6 (Odds Only)
34. Harmonic Enhancer Sweet Spot
35. High-End (Triode) Audio Pre-Amplifier
36. Overloaded Guitar Power Amplifier
37. Overloaded Guitar Pre-Amplifier
38. Pentode Microphone Pre-Amplifier 1
39. Pentode Microphone Pre-Amplifier 2
40. Purist 1
41. Purist 2
42. Purist 3
43. Red Hot Jazz
44. Triode Tube Pre-Amplifier 1
45. Triode Tube Pre-Amplifier 2
46. Triode Tube Pre-Amplifier 3
47. Triode Tube Pre-Amplifier 4
48. Triode Tube Pre-Amplifier 5
49. Triode Tube Pre-Amplifier 6

50. Vacuum Tube Fuzz Box 1
51. Vacuum Tube Fuzz Box 2
52. Vacuum Tube Fuzz Box 3
53. Vacuum Tube Fuzz Box 4

Voice Gerbler

1. Default
2. Female Child Downshift 1
3. Female Child Downshift 2
4. Female Voice Downshift 1
5. Female Voice Downshift 2
6. Female Voice Downshift 3
7. Female Voice Upshift 1
8. Female Voice Upshift 2
9. Male Child Downshift 1
10. Male Child Downshift 2
11. Male Child Upshift 1
12. Male Child Upshift 2
13. Male Voice Downshift 1
14. Male Voice Downshift 2
15. Male Voice Upshift 1
16. Male Voice Upshift 2
17. Silly Girl

VVA Presets (Relating to the Fat Bass Feature)

1. Fat Bass – Audiophile Quality
2. Fat Bass – B15 (Ampeg)
3. Fat Bass – Balanced Harmonics
4. Fat Bass – Boomy
5. Fat Bass – Boomy 2
6. Fat Bass – Compressed & Grungy
7. Fat Bass – Even & Odd Harmonics
8. Fat Bass – Grimey
9. Fat Bass – Grungy
10. Fat Bass – Odd Harmonics
11. Fat Bass – Round Bottom End
12. Fat Bass – Subtle Effect

A Brief History of Diamond Cut Productions

In the spring of 1986, an R&D engineer/scientist by the name of Craig Maier read an article in The Star Ledger, a local newspaper, entitled "Budget Cuts Cast Shadow on Edison National Historic Site." The article, written by science editor Kitta McPherson, described the deteriorating condition of the Edison National Historic Site and its archives located in West Orange, New Jersey. Among the many artifacts which were not receiving the proper curatorial attention due to poor funding was a collection of test-press recordings which were made by the Edison Company between the years of 1927 through 1929, which was their last few years in the record business. Craig told a friend and fellow engineer named Rick Carlson about the article in hopes that it might stir up in him some interest in the Edison site as well. Craig and Rick, after some considerable discussion, decided to offer to volunteer some of their spare time and technical expertise in the area of audio hardware and software engineering in order that the Edison Lateral collection of test pressing recordings could be transferred to digital tape so that the "sound artifacts" would be eternally preserved and archived in the digital domain at the site.

Contact was made with then Supervisor Museum Curator, Dr. Edward Pershey, Ph.D. During their first meeting at the site, Dr. Pershey showed the two engineers thousands of one-of-a-kind test pressing recordings that were piled in stacks on a long row of tables on the second floor of the Edison main laboratory building. This initial introduction to the collection was an earnest attempt to sober up these two individuals as to the magnitude of the undertaking for which they were volunteering. The total number of songs which were recorded numbered over 1200 in anywhere from two to five takes each. This only further increased their interest in the project since the possibility of finding some truly important music that had previously been unheard since the late 1920's would be quite high in such a large collection of test pressings. After several additional meetings with Dr. Pershey, an informal agreement was made such that the two engineers could proceed to seek out funding from private sources to set up an audio restoration laboratory in one of their own homes for the project. They contacted around 30 companies in the New Jersey area seeking funds to help build their laboratories. After about seven months of effort, they succeeded in raising enough money to fund their project. In addition to fund raising, they also designed and constructed several pieces of custom equipment that was needed for the project (equipment which was not readily available on the market at the time).

The next step was to become educated in the proper technique of archival audio transferring. To that end, they hired Mr. Tom Owens of the Rogers and Hammerstein musical library in New York City as an engineering consultant. Tom spent time with the two engineers at the New York City Public Library sound lab (Rogers and Hammerstein) teaching them some of the "tricks of the trade." Tom also visited the first sound lab which the two engineers set up for the restoration project located at Craig's home in Verona, NJ. He provided constructive criticism regarding the sound lab which the two engineers had set up, allowing them to improve upon their initial system. One significant problem which Tom highlighted for the two engineers was that of establishing the correct turnover frequency for the transfer of these lateral test pressings. Documentation could not be found at the Edison site regarding the specifics of this important parameter. So Rick and Craig devised some experiments which were conducted on a "high-end" vacuum tube based Edison phonograph designed around the same time period as the test pressings in order to deduce the correct turnover frequency. After their experiments,

modifications were made to their magnetic phonograph pre-amplifier to provide the most likely proper turnover frequency for the transfers.

A seven year pro-bono contract was drawn up between the Edison National Historic Site / U.S. Department of the Interior, and Rick Carlson and Craig Maier for the purposes of executing the project outlined above. Finally, the two engineers were ready to begin the project. Nearly one full year had lapsed before the first record was transferred to digital tape at Craig's home in Verona, N.J. Shortly thereafter, the sound lab was rebuilt in the Maier's new home in Rockaway Township, NJ. That is the location in which the lion's share of the transfer project took place over the next seven years.

After transferring around 900 of the songs (times 2 - 5 takes per song, about 2,200 transfers in total) Craig and Rick decided that the music was not doing much good sitting in the underground vault of a museum. Since they were the only two people alive who had heard almost the entire collection, they decided that it would be a good idea to try to release some of this previously unreleased material (only around 200 of the songs had ever been released in the Edison lateral format). So they approached the Edison site in order to try to accomplish this. After about one year of frustration in dealing with the bureaucracy, they decided it would be a lot easier to form their own company and release these songs under their own record label. Thus was formed Diamond Cut Productions in 1992 with Craig and Rick providing their own seed capital for the venture. Their first release entitled "Unreleased Edison Laterals 1 - - - an anthology of Edison Needle type records" was such a success in the market that they were able to start another project in 1994 entitled "The California Ramblers, Edison Laterals 2." For this project, they decided to improve on the audio restoration process, which they had used on their previous release. Instead of analog signal processing, they migrated to digital signal processing utilizing their own algorithms to remove crackle, ticks, pops and hiss from the original material. They named their process (which ran on an inexpensive pc) "Diamond Cut Audio restoration tools" or DC-Art for short. Their technique proved successful to the extent that the Smithsonian Institution Press employed Diamond Cut Productions to perform audio restoration for some of their American Songwriter Series of CD releases using this process. Diamond Cut's third CD release entitled "Hot Dance of the Roaring 20's, Edison Laterals 3" was processed utilizing exclusively their own audio restoration program; all analog processing equipment had been abandoned by this point in time. In the meantime and in parallel with the efforts to bring "Hot Dance . . ." to the market, Craig worked with County records to produce and release an Edison olde time group on CD called "Ernest Stoneman and his Dixie Mountaineers" using their audio restoration process. In the spring of 1996, their program was first formally introduced into the commercial marketplace at a meeting of "Record Research" which was held at the Maier residence in Rockaway Township, NJ. Since then it has been sold throughout the world for not only musical audio restoration applications, but for others such as 911 call restoration, clarification of police surveillance recordings, cleanup of radio broadcasts for release on CD, restoration of historic spoken word recordings, cockpit voice recording restoration, plus many others.

Diamond Cut Productions, Inc. has now become one of the predominant international players in the audio restoration and enhancement software market. New features and improved performance will be added into their legacy audio restoration software products on a continuous process basis.

In the future, Diamond Cut Productions expects to continue releasing more CDs in their Edison Lateral Cut series. However, they have also branched out into other musical venues from the 1920's and 1930's time period.

Diamond Cut Audio Restoration Tools Development Timeline

Mid 1993 -

While frustratingly removing clicks from an Edison record using just a computer and a mouse whilst drinking too much wine and beer, Rick Carlson and Craig Maier wrote the first few lines of code in an attempt to perform this process automatically. Although very crude at the beginning, this particular algorithm (the Impulse Filter) eventually developed into its present level of sophistication occupying about 500 K of compiled code. Several of those original lines and thinking are still in the present day code. Rick had stored those original lines of code under the filename of dcart.exe ultimately giving rise to the name "Diamond Cut audio restoration tools."

Late 1993 -

Having heard about Diamond Cut through the grapevine, Bruce Talbot, executive producer for the Smithsonian Collection of Recordings contacted us and inquired if they would be willing to apply their "de-clicking" process to a particularly bad recording of "Darn That Dream." This restoration was released on the American Songbook Series on a CD entitled "James Van Heusen." which was copyrighted in 1994 as release number RD 048-18 / A 23955.

Early 1994 -

Having been satisfied with their work on the "James Van Heusen" CD, Bruce Talbot again contacted Diamond Cut to do some work on a release in the American Songbook Series, which was to be called "Richard Whiting." On it, they restored a "basket case" copy of a Rudy Vallée rendition of "Honey." In the liner notes for the CD it says "The only source available for "Honey" was a 78-rpm disc in very poor condition. The sound quality has been greatly improved by Craig Maier and Rick Carlson of Diamond Cut Productions, using their DC-Art system of sound restoration." This CD is also copyrighted in 1994 but under the release number of RD048-22 / A 24571.

Mid 1994 -

While recuperating from surgery and being extremely bored, Craig decided to restore an entire artist's output of Edison Lateral Cut records which Rick and Craig had earlier transferred to digital audiotape as part of the Edison Lateral Cut Disc archival project. The artist was Ernest V. Stoneman (and his Dixie Mountaineers). He used their DC-Art program to remove all extraneous clicks, pops, and hiss from the originals. Ultimately, Diamond Cut Productions sold the Master digital tape to County Records which released and copyrighted the CD in 1996 on County release number CD-3510.

Late 1994 -

Diamond Cut Productions decided that it was time to release another in its series of mostly Unreleased Edison Lateral Cut recordings. They choose "The California Ramblers" as the subject of the release. They also decided to use no additional analog processing equipment other than their DC-Art program to restore the recording. De-clicking, de-popping, de-hissing, de-rumbling and minor equalization were all performed via their computer algorithms. Thus was their first release which included only two analog steps, that of the mastering back in the

late 1920's and that of the transfer to digital tape in the late 1980's and early 1990's. This CD is entitled "The California Ramblers - - - Edison Laterals 2" and was released under number DCP-301D and copyrighted in late 1994. This CD is still available from Diamond Cut Productions, Inc. and other venues throughout the world.

Early 1995 -

Diamond Cut Productions decided to expand their business from the production of CD's to the manufacture of audio restoration software products. They decided that since the program had been quite useful to in their CD business, it might also be useful to many other people with audio collections in need of restoration. The first public release took about 7 months to smooth out the bumps in the program that we had learned to work around. Making a commercial software product is much different than writing one for yourself as they were soon to find out!

April 21st 1995 -

Diamond Cut Productions sent out the initial Beta version of DC-Art to some potential customers identified as QA 1.1. They worked on de-bugging this software for the next few months. If you have an original copy of QA 1.1, it may be an antique! (Antique software - - - what's this world coming to??)

July 1996 -

Version 1.0 was officially released at a meeting of "Record Research" held at the Maier Rockaway Township residence. The first copies were sold at the end of the meeting during which the program had been demonstrated to a small group of about 15 people.

September 4th, 1997 -

A distributorship agreement is signed between Diamond Cut Productions, Inc. and Tracer Technologies, Inc. in order to help facilitate the marketing and distribution of the DC-Art product line.

December 1997 -

Version 2.0 of DC-Art featuring, among other things, real time Preview was introduced into the market.

September 1998 -

Version 3.0 of DC-Art, otherwise known as Diamond Cut 32 was introduced into the market with novel features such as the Virtual Valve Amplifier.

August 1999 -

Version 4.0 otherwise known as Live and Millennium was introduced. This brought with it a new level of performance and features in the audio restoration and enhancement software market. Features like Live feed through mode, and 24-bit/96 kHz support have changed the landscape in this area of endeavor.

December 1999 -

Established German distributorship for the Live and Millennium programs through Digital Broadcast Systems GmbH (dBS). The products can be seen and purchased in the German language at www.diamondcut.de

August 2001 -

Updated Live and Millennium to version 4.8 with bug fixes and the addition of a high resolution VU Meter. Also, the frequency resolution of the spectrum analyzer was greatly enhanced.

February 2001 -

Released the code for Enhance/MP3, a low cost product used for improving the sound quality of MP3 audio files. Diamond Cut Productions Inc. and Tracer Technologies form a new Partnership called Enhanced Audio Inc., whose sole purpose is to market, and distribute the world's best audio enhancement, restoration, and analysis tools.

June 2002 -

Released Live/Five to Beta

September 2002 -

Released Live/Five to Production

May 2004 -

Released Live/Six to Beta

August 2004 -

Released Live/Six to Production

July 2007 -

Released Version 7 to Beta

October 2007 -

Released Version 7 to Production

November 2008 -

Released Live/Forensics Audio Laboratory, Version 7.5 to Beta

January 2009 -

Released Live/Forensics Version Audio Laboratory, Version 7.5 to Production

November 2009 -

Released DC8, Version 8.0 to Beta

March 2010 -

Released DC8, Version 8.0 to Production

November 2010 -

Released DC Forensics8 Audio Laboratory, Version 8.0 to Beta

April 2011 -

Released DC Forensics8 Audio Laboratory, Version 8.0 to Production

December 2011 -

Released DC Forensics8 Audio Laboratory, Version 8.1 to Production

April 2012 –
Released DC8, Version 8.1 to Production

Diamond Cut Productions Edison Lateral Series CD and Cassette Releases

Diamond Cut Productions is proud to offer releases of Edison Lateral Cut Test Pressing recordings available in the CD format, with one available in both CD and Cassette. Also, we have an assortment of non-Edison recordings available. The following is our current product offering:

1. Unreleased Edison Laterals 1 CD Version (DCP-201D) - - - \$17.95
Unreleased Edison Laterals 1 Cassette Version (DCP-201S) - - - \$9.95
2. The California Ramblers, Edison Laterals 2 (DCP-301D) - - - \$17.95
3. Hot Dance of the Roaring 20's, Edison Laterals 3 (DCP-202D) - - - \$17.95
4. Ernest V. Stoneman and his Dixie Mountaineers (Edison) - - - \$13.95
5. Eva Taylor with Clarence Williams, Edison Laterals 4 (DCP-303D) - - - \$17.95
6. Vaughn De Leath - The Original Radio Girl, Edison Laterals 5 (DCP-304D) - - - \$17.95
7. Hot and Rare - Hot Dance tunes from Rare Jazz Recordings (DCP-203D) - - - \$17.95
8. B.A. Rolfe and his Lucky Strike Orchestra (DCP-305D) - - - \$17.95
9. The Marvelous Melodies of Peter Mendoza (DCP-306D) - - - \$17.95
10. Edison Diamond Disc Fox Trots: 1920 - 1923 (DCP-307D) - - - \$17.95
11. Rudy Vallée and His Connecticut Yankees: 1928 - 1930 (DCP-308D) - - - \$17.95
12. Eddie Duchin and his Central Park Casino Orchestra 1932 - 1937 (DCP-309D) - - - \$17.95
13. Jazz: It's a Wonderful Sound! (DCP-500D) - - - \$17.95
14. Ray Noble Plays Ray Noble (and others) 1935 - 1950 - - - \$17.95 each

For all other items, please include \$2.00 per item to cover Shipping and Handling in the U.S. Europe, please include \$4.00 per item.
N.J residents add 7% Sales Tax.

To order, please visit our on-line store located at www.diamondcut.com, or please send your order including your check or money order payable in U.S funds to:

Diamond Cut Productions, Inc.
P.O. Box 305
Hibernia, NJ 07842-0305

Fax your order to 973-316-5098
Visa, MasterCard or Discover Accepted
Call in your order at 973-316-9111

Contact us via our website or make your purchase at our online store
located at www.diamondcut.com

Note: Dealers, Retailers, and Record Distributors are invited to request a
copy of our Product Discount Schedule.

DCAT-3 Audio Test CD Set



DCAT-3 is a comprehensive set of 147 audio signals contained on a 3 CD set, which is very useful for room acoustical response balancing and audio equipment testing and evaluation. Room acoustical response balancing can be performed with nothing more than a calibrated low cost sound pressure meter in conjunction with the DCAT-3's set of 31 narrowband random $1/3^{\text{rd}}$ octave weighted noise signals.

All DCAT-3 signals are announced on a separate track allowing any signal to be repeated using the loop-play feature on your CD player. DCAT-3's signals are all tightly calibrated in frequency, amplitude and spectral distribution since they are all mathematically synthesized without the use of A/D converters. Sine waves are very low in harmonic distortion content while square waves exhibit excellent rise times while triangle waves display exceptional linearity. Also, none of the signals contain any annoying transients since each signal is smoothly faded-in and faded-out. DCAT-3 offers the user a great value at a very reasonable price. DCAT-3 contains the following signal types:

- Discrete Sine, Square, Triangle Waveforms
- Swept Sine, Square and Triangle Waveforms

- Random Noise signals including White, Pink, Brown, Subsonic and Seismic
- Weighted Narrowband Random noise in 1/3rd Octave increments (31 signals)
- Phase inverted and Quadrature signals
- Left & Right Channel Only signals
- Dual Tone signal for Intermodulation Testing
- Calibrated silence (DAC counts = 0)

DCAT-3 Performance Specifications

Signal Type	Frequency Accuracy	Amplitude Accuracy	Distortion (% THD)
Sine	+ / - 0.005 %	+ / - 0.5 dB	0.0033 % @ 1 kHz
Square	+ / - 0.005 %	+ / - 0.5 dB	44 % @ 1 kHz
Triangle	+ / - 0.008 %	+ / - 0.5 dB	12 % @ 1 kHz
White	N/A	+ / - 0.7 dB	100 %
Pink	N/A	+ / - 1.5 dB	100 %
Narrowband Random (1/3 rd Octave Steps)	+ / - 0.2 %	+ / - 1.5 dB	100 %
Swept Sine	+ / - 0.005 %	+ / - 0.5 dB	0.0033 % @ 1 kHz
Swept Square	+ / - 0.005 %	+ / - 0.5 dB	44 % @ 1 kHz
Swept Triangle	+ / - 0.5 %	+ / - 1.5 dB	12 % @ 1 kHz
Dual Tone	+ / - 0.005 %	+ / - 0.5 dB	0.008 % (high tone)

*The quality of your CD player's performance may degrade these numbers to varying degrees

To Order, just go to www.diamondcut.com and follow the signs to the online store.

Tracer Technologies...For the Supplies You Need

Tracer Technologies has been in the business of supplying audio products for the last 14 years. If you need a few little tweaks to finish your system, pay us a visit at www.tracertek.com

- **Sound Cards:** The most important part of this equation. You now own the world's best audio restoration software...don't funnel it through anything that has the word "Blaster" in it...or any bundled sound card.
- **Stereo Preamp:** You cannot plug a turntable directly into your sound card unless it is equipped with a digital output port. If you don't want to drag your whole stereo to your computer, Tracer offers a full assortment of preamps that will make this job cake.
- **Speakers:** Make your ears happy. Stop depending on those cheesy little bundled speakers that came with your computer.
- **Headphones:** All part of the ear happiness theory. Work on your audio files while the rest of the house sleeps. Also hear greater detail while you work.
- **Mixers:** Any easy way to plug anything into your sound card. Mics, instruments, etc. quickly and easily plug in.
- **Microphones:** If you've got to record your voice...stop looking, Tracer has a line of Mics that are inexpensive, but rivals to the world's best studio Mics.
- **Multi-track Software:** The one thing that EIGHT/DC FORENSICS doesn't do. Now you can quickly and easily record multiple tracks just like a professional recording studio.
- **Record Cleaning Kits:** Before you restore; clean. These kits can help save needle and record wear by giving you a clean surface to start.
- **Turntables:** If you're in the market for a good turntable, we have a line that is very high quality, but won't cost an arm and a leg.
- **MP3 Players:** Now that you can make your own MP3s, why not try our cool little portable MP3 players.

Tracer Technologies
P.O. Box 189
Windsor, PA
17366
tel: 717-764-9240

Diamond Cut Software Product Model Number Nomenclature
(English Versions)

DC8, v. 8.1 – Full Version: **DCP-908F**

DC8, v. 8.1 – Upgraded Version: **DCP-908UP**

DC Forensics8 Audio Laboratory, v. 8.1 – Full Version: **DCP-918F**

DC Forensics8 Audio Laboratory, v. 8.1 – Upgraded Version: **DCP-918UP**

License Agreement

This is a legal agreement between you, the end user, and Diamond Cut Productions, Inc. (Diamond Cut Productions). The enclosed Diamond Cut Productions software program (the SOFTWARE) is licensed by Diamond Cut Productions for use only on the terms set forth herein. Please read this license agreement. Installing this software indicates that you accept these terms. If you do not agree to these terms, please contact Diamond Cut Productions or its representatives within 30 days.

GRANT OF LICENSE. Diamond Cut Productions grants to you the right to use one copy of the enclosed DCEIGHT/DC FORENSICS SOFTWARE on a single terminal connected to a single computer (i.e. single CPU) or to a network server. If you install the SOFTWARE on a network server, you must purchase a separate copy of the SOFTWARE for each computer terminal that will be used to operate the SOFTWARE.

GRANT OF NETWORK LICENSE. If you are acquiring a version of the SOFTWARE specifically intended for network use, Diamond Cut Productions grants you the right to use the SOFTWARE on a LICENSED COMPUTER NETWORK as provided below. A computer network is any combination of two or more terminals that are electronically linked and capable of sharing the use of a single software program. A LICENSED COMPUTER NETWORK is a computer network for which you have acquired and dedicated at least one (1) Diamond Cut Productions standard version of the SOFTWARE (which can run stand-alone or as a network server). For additional users to use the SOFTWARE on the network you must acquire a WRITTEN LICENSING AGREEMENT from Diamond Cut Productions indicating the number of users expected to be able to use the SOFTWARE. You may have as many copies of the SOFTWARE in simultaneous use on the network as is specifically authorized in the WRITTEN LICENSING AGREEMENT.

COPYRIGHT. The SOFTWARE is owned by Diamond Cut Productions and is protected by United States copyright laws and international treaty provisions. You may either (a) make one copy of the SOFTWARE solely for backup or archival purposes provided that you reproduce all copyright and other proprietary notices that are on the original copy of the SOFTWARE provided to you, or (b) transfer the SOFTWARE to a single hard disk provided you keep the original solely for backup or archival purposes.

OTHER RESTRICTIONS. You may not rent or lease the SOFTWARE. You may not reverse engineer, decompile, disassemble, or create derivative works from the SOFTWARE.

GOVERNMENT LICENSEE. If you are acquiring the SOFTWARE on behalf of any unit or agency of the United States Government, the following provisions apply:

The Government acknowledges Diamond Cut Productions representation that the SOFTWARE and its documentation were developed at private expense and no part of them is in the public domain.

The Government acknowledges Diamond Cut Production's representation that the SOFTWARE is "Restricted Computer Software" as that term is defined in Clause 52.227-19 of the Federal Acquisition Regulations (FAR) and is "Commercial Computer Software" as that term is defined in Subpart 227.471 of the Department of Defense Federal Acquisition Regulation Supplement (DFARS). The Government agrees that:

(i) If the SOFTWARE is supplied to the Department of Defense (DoD), the SOFTWARE is classified as "Commercial Computer Software" and the Government is acquiring only "restricted rights" in the SOFTWARE and its documentation as that term is defined in Clause 252.227-7013 (c) (1) of the DFARS, and

(ii) If the SOFTWARE is supplied to any unit or agency of the United States Government other than DoD, the Government's rights in the SOFTWARE and its documentation will be as defined in Clause 52.227-19 (c) (2) of the FAR.

RESTRICTED RIGHTS LEGEND. Use, duplication or disclosure by the Government is subject to restrictions as set forth in subparagraph (c) (1) (ii) of the Rights in Technical Data and Computer Software clause at DFARS 252.227-7013. Diamond Cut Productions, P.O. Box 305, Hibernia, NJ 07842-0305.

EXPORT LAW ASSURANCES. You acknowledge and agree that the SOFTWARE is subject to restrictions and controls imposed by the United States Export Administration (the "Act") and the regulations thereunder. You agree and certify that neither the SOFTWARE nor any direct product thereof is being or will be acquired, shipped, transferred or re-exported, directly or indirectly, into any country prohibited by the Act and the regulations thereunder or will be used for any purposes prohibited by the same.

GENERAL. This Agreement will be governed by the laws of the State of New Jersey, except for that body of law dealing with conflicts of law. Should you have any questions concerning this Agreement, or if you desire to contact Diamond Cut Productions for any reason, please write: Diamond Cut Productions Customer Sales and Service, P.O. Box 305, Hibernia, NJ 07842-0305.

NUMBER OF COPIES LICENSED. If you have not purchased a license that authorizes use of the Software on multiple computers or by multiple individuals, then you are authorized to use ONLY a single copy of the Software on a single computer. Only ONE copy of the Software may be created for archival or backup purposes. All copies of the Software MUST include the Diamond Cut copyright notice and other legal notices.

TERM. This license is effective from your date of purchase and shall remain in force until terminated. You may terminate the license and this License Agreement at any time by destroying the Software and the accompanying documentation, together with all copies in any form.

LIMITED WARRANTY. Diamond Cut Productions, Inc. warrants that the SOFTWARE will perform substantially in accordance with the accompanying written materials for a period of ninety (90) days from the date of receipt. Any implied warranties on the SOFTWARE are limited to ninety (90) days. Some states do not allow limitations on duration of an implied warranty, so the above limitation may not apply to you.

CUSTOMER REMEDIES. Diamond Cut Productions, Inc. entire liability and your exclusive remedy shall be, at Diamond Cut Productions option, either (a) return of the price paid or (b) repair or replacement of the SOFTWARE that does not meet Diamond Cut Productions Limited Warranty and that is returned to Diamond Cut Productions with a copy of your receipt. This Limited Warranty is void if failure of the SOFTWARE has resulted from accident, abuse, or misapplication. Any replacement SOFTWARE will be warranted for the remainder of the original warranty period or thirty (30) days, whichever is longer. These remedies are not available outside of the United States of America.

NO OTHER WARRANTIES. Diamond Cut Productions, Inc. disclaims all other warranties, either express or implied, including but not limited to implied warranties of merchantability and fitness for a particular purpose, with respect to the SOFTWARE and any accompanying written materials. This limited warranty gives you specific legal rights. You may have others, which vary from state to state.

NO LIABILITY FOR CONSEQUENTIAL DAMAGES. In no event shall Diamond Cut Productions, Inc. or its suppliers be liable for any damages whatsoever (including, without limitation, damages for loss of business profits, business interruption, loss of business information, or other pecuniary loss) arising out of the use of or inability to use this Diamond Cut Productions, Inc. product, even if Diamond Cut Productions, Inc. has been advised of the possibility of such damages. Because some states do not allow the exclusion or limitation of liability for consequential or incidental damages the above limitation may not apply to you.

PRODUCT CHANGES. Product features and specifications are subject to change without notice.

Index

- %THD, 356
- .aif, 74
- .asf files, 71
- .avi files, 71
- .bak, 83
- .csv
 - Formants Export, 320
- .cue, 90
 - export this format, 91
 - import this format, 90
- .flac
 - FLAC, 67
- .m3u
 - export this format, 91
 - import this format, 90
- .MP3, 94
- .mpeg
 - .mpeg files, 71
- .mpg
 - .mpg files, 71
- .ogg, 74
- .pkf, 381
- .pls
 - export this format, 91
- .ses files, 371
- .spt, 364
- .txt, 74
 - Formants Export, 320
- .wav file, 28
- .wls
 - import this format, 90
- .WMA File Encoding, 91
- .zip
 - presets, 152
- “Q” point, 254
- 10 Band Graphic Equalizer, 224
- 12AT7**, 259
- 12AU7, 259
- 12AX7, 255, 258
- 20 Band Graphic Equalizer, 226
- 2A3**, 260
- 2SK-175 MOSFET**, 261
- 3 Band Equalizer, 239
- 30 Band Graphic Equalizer, 227
- 32,000 Band Graphic Equalizer, 300
- 5881, 260
- 6267 / EF86, 261
- 6EJ7**, 259
- 6L6GC, 259
- About Dcart, 385
- Accelerators, 513
- acetate, 391
- Acetate Tape, 391
- acetone, 422
- Acknowledgements
 - Credits, 4
- acoustic feedback, 442
- acoustically mastered, 244
- Adaptation Speed**, 294
- adaptive
 - Auto Voice Filter, 334
- adaptive equalizer, 300
- Adaptive Filter, 292
- Adaptive Frequency Domain Filter, 195
- ADPCM**, 459
- AES, 459
 - LP EQ Curve, 245
- AGC, 266
- Aiff, 68
- A-law Compression**, 459
- Album
 - DCTune Library, 81
- ALC, 266
- alcohol, 422

- algorithm, 431
- Aliasing artifacts, 196
- Alternate FFT
 - Preference, 146
- ambience, 268
- ambient sound, 314
- American 78's, 243
- Amphenol, 494
- amplifier, 23
- amplitude, 29
- Analog Noise Filter, 201
- Analog Tape Hiss**, 392
- Analog tape recording, 390
- Append to End, 108
- Application Notes, 492
- Archival Recording, 395
- archive, 80
- Aromatic solvents, 422
- Arrange Icons, 382
- articulation, 313
- Artifact Suppression Mode**, 190
- Artist
 - DCTune Library, 81
- Attack Time**, 188
- Attenuate**, 460
- Attenuation chart, 493
- attenuator, 493
- audible enunciator, 147
- Audible Wave Analyzer**, 443
- Audio Connection Standards, 494
- Audio Extraction, 71
- Audio Frequency Spectrum, 496
- audio products, 557
- Auto EQ, 300
- Auto Leveling, 157
- Auto Sample**, 306
- Auto Spectrum
 - auto spectrum CNF, 186
 - Auto Spectrum CNF Mode, 185, 193
 - Auto Voice Filter
 - forensic, 334
 - automatic De-Clipping**, 401
 - automatic equalizer, 300
 - Automatic Level Control, 266
 - Available Hard Drive Space**, 381
 - Available Recording Time**, 133
 - Average Angle, 367
 - averaged vector displacement
 - angle, 366
 - Averaging Filter, 222
 - azimuth, 366
 - Azimuth Correction, 235
 - Background Tasks, 27
 - Backup First**, 421
 - Balance control, 241
 - Band Pass Filter, 208
 - Bar Graphs**, 172
 - bass, 224
 - Batch File Editor, 154, 157
 - BBS, 492
 - BCF
 - Big Click Filter, 182
 - beam power pentode, 259
 - Berliner**, 461
 - Bessel, 356, 365
 - Big Click Filter, 182
 - bi-modal interpolation
 - time & frequency domain, 110
 - binaural, 293
 - bit depth conversion, 143
 - Bit Rate**, 380
 - Blackman, 354
 - bleep
 - tone, 114
 - blend control, 269
 - Blend to Mono, 421

- Blue Amberol**, 427
- Box Zooming
 - with rectangle, 375
- brass instrument, 172
- Brick Wall Filter, 296
- brilliance, 285
- Broadcast .wav
 - supported format, 67
- Broadcast Wave
 - file header editing, 127
- Brown Noise, 139
- Buffers, 147
- bug fixes, 12
- Burn Options, 347
- Butterworth, 209
- Butterworth Filter**, 462
- Buzz**, 462
- BWF
 - Broadcast Wave, 127
- Bypass, 40
- cadence, 277
- Cancel**, 42
- capstan, 275
- cascade
 - filters in Multifilter, 161
- Cascade, 382
- Cassette**, 392
- CBR, 149
- CD burner, 347
- CD Player
 - DCTune Library, 94
- CD Prep Menu, 341
- CD Ripper**, 95
- CDDB Lookup
 - cd database lookup, 96
- CDDB Setup**, 150
- CDMA Phones, 407
- CDR Prep, 396
- cell phone noise, 406
- Cell Phone Noise Filter
 - Forensics, 332
- cepstrum
 - Voice ID, 317
- Ceramic Phono Cartridges, 25
- Chain of Custody
 - Forensics Recordings, 409
- Change Resolution, 142
- Change Sample Rate, 142
- Change Speed, 273
- channel balance, 230
- Channel Blender, 267
- Channels**, 131
- Chebyshev Filter**, 463
- Chimes, 147
- chirp Z transforms, 358
- Choosing between Classic and
 - Fast Editing mode, 31
- Chop File into Pieces, 342
- Chroma/Intensity Modulation,
 - 312
- cine, 309
- Class AB, 463
- class-A, 260
- Classic Editing mode, 345
- Classic mode**, 450
- Classification of Amplifiers**,
 - 463
- Clean Display, 148
- Clear All Markers, 340
- Clear Button**, 351
- Clicks, 170
- Clip Level**
 - De-Clipper, 324
- Clipping**, 464
- Clone Source**, 100
- Close All, 382
- Close Source, 94
- closed form
 - VPA Math Method, 238
- CNF, 184

- co-axial cables, 418
- Color Palette**, 312
- Columbia, 424
- Columbia LP curve, 243
- comb, 198
- Comb Filter, 403
- Compact Disc**, 463
- complex cepstrum
 - Voice ID, 317
- Compressor, 264, 270
- Concatenate File, 157
- constant velocity, 417
- Contact Information, 458
- Context Sensitive Help, 384
- Continuous Noise, 182, 184
- Continuous Playback Mode
 - DCTune Library, 88
- Control Points, 46
- Convergence**, 294
- Conversion Quality, 142
- Copy, 105
- Copy and Paste, 105
- Copyright, 3, 385
- Corner Frequency**, 465
- Crackle, 170, 171
- create .mp3s, 74
- Cross Fade, 237
- Crossfade, 112
- Cross-fade, 237
- crossover frequency, 302
- CUE
 - export file format, 91
 - Import File Format, 90
- cue words, 313
 - voice print, 313
- Current System Status**, 133
- cursor, 136
- curvilinear**, 275
- Custom Sample Rate, 132
- Cut, 107
- cylinder, 395
- DAO, 347
- DAT, 26, 471
- Data Disk Burner, 75
- database, 80
- db/Octave, 431
- DC Offset**, 466
- DC Tune Library, 80
- DCArt
 - Diamond Cut Audio
 - Restoration Tools, 12
- DCAT-3 Audio Test CD Set, 555
- DDD, 253
- decay, 264
- Decibels, 498
- decimation, 358
- Decimation, 315
- De-click**, 51
- Declicking, 400
- DeClipper, 322
- De-Crackling, 221
- deep bass, 283
- De-Esser, 265, 266
- delay lines, 250
- delay time, 264
- Delete All Temp Files, 124
- Deleting Wave files, 101
- demo, 383
- Demo .wav Files, 386
- Destination window, 55
- detail control, 256
- Development Timeline, 549
- Device I/O Selection, 136
- dial presence, 270
- Dial Tone Phone Frequency
 - Chart, 499
- Diamond Cut Productions, 466
- Diamond Cut Productions, Inc
 - Home Page, 492
- Diamond Discs**, 466

- Digital Rights Management, 81
- DIN, 495
- Direct Spectral Editing, 115
- DirectX Filters, 167
- Disc At Once**, 347
- Disc Space Consumption**, 512
- Disguised Voice Effect, 279
- Display Colors, 148
- Display Controls**, 310
- Display Frequency Labels**, 313
- Display Frequency Range**, 313
- Display Mode**, 350
- Display Preferences, 148
- Display Time Format, 148
- Display X-Axis, 148
- Distortion, 355
- Distortion Analyzer, 355
- Dither, 143, 467
- double precision floating point
 - math, 15
- Drive**, 467
- Drive & Directory, 144
- DRM, 67
 - Digital Rights Management, 67
- Drop A Marker, 340
- drop marker, 340
- Dry, 252
- dry run**, 423
- DSE
 - Direct Spectral Editing, 115
- DSS™, 325
- DTMF(touch tone) signals, 402
- Dual Waveform Display**, 350
- dwelt-time, 250
- Dynamic Noise Filter, 201
- dynamic range, 263
- Dynamic Rumble (Only) Filter
 - CNF Preset, 198

- Dynamic Spectral Subtraction™, 325
- Dynamics Processor, 262
- Early Columbia LP's, 243
- ECC83, 258
- Echo, 250
- echo chamber, 250
- Echo Effect, 250
- Edge Sharpness**
 - DSE, 117
- Edison Lateral Series, 553
- Edit History*, 145
- Edit Menu, 104
- EF183, 259
- Effects Menu, 248
- electromagnetic field, 416
- Electron Tube**, 468
- EQ Matching, 301
- Equalization chart for LP
 - records, 525
- equalization curves, 230
- equal-tempered scale, 140
 - musical, 140
- Erase**
 - Direct Spectral Editor**, 116
 - DSE, 116
 - European 78's, 244
- Exciter, 255
- Exciter Control**, 287
- Expanded File Conversion**
 - (Large Size File Conversion), 69
- Expander, 263, 270
- Expert Impulse filter, 170
- Expert Impulse Noise, 174
- Exponential, 282
- Export
 - playlists, 91
- Extended Recording, 134
- Extent Control**

- DSE, 117
- EZ Clean Filter, 158
- EZ Enhancer, 286
- EZ Forensics Filter, 290
- EZ Impulse Noise, 170
- Fade In, 122
- Fade Out, 123
- FAQs
 - Frequently Asked Questions, 448
- Fast Edit History, 371
- Fast Forward, 129
- Fast-Edit (single file editing mode) mode, 27
- Fat Bass, 257
- feedback, 215
- Feedback, 251
- feedback control, 252
- feed-through
 - real-time, 161
- FFRR, 469
 - LP EQ Curve, 245
- FFT, 187
- FFT analyzer, 358
- FFT Size (Resolution)**, 192
- FFT Size(number of Frequency Bands)**, 305
- FIFO**, 469
- File Conversion, 230
- file header
 - editing, 126
- File Info, 381
- File Menu, 67
- File Naming**, 344
- file paths**, 457
 - Forensics8, 457
- File Properties
 - edit menu, 126
- File Split and Recombine, 151
- Filter Co-efficients, 295
- Filter Finder**, 61
 - filter frequency, 445
- Filter Length**, 294
- Filter Menu, 154
- Filter Slopes, 210
- Filter Sweeper, 280
- Find and Mark
 - Silent Passages, 344
- Find and Mark Silent Passages, 344
- FIR, 296
- Firewire
 - Ripping to, 97
- FLAC
 - Lossless Compression, 446
- Flashback, 166
 - memory, 166
- flat phono pre-amp, 239
- flat phono preamplifier, 239
- Flat Spectrum
 - dither, 143
- Fletcher-**, 469, 478
- Fletcher-Munson, 469
- Flutter**, 469
- FM stereo, 268
- font size, 104
- Forensic Tape Authentication, 403
- Forensics AFDF, 195
- Forensics AFDF Mode, 185
- Forensics Menu, 288
- formants
 - vocal, 313
 - Voice ID, 317
- Forum, 492
- Free Disk Space**, 133
- Freeze button**, 294
- Frequency Axis Selector**, 312
 - frequency bands, 352
 - frequency bins, 362

frequency domain interpolation,
111

Frequency Labels, 313

Frequency Resolution of

Spectrum Analyzer, 351

frequency response, 391

Voice ID, 317

full duplex, 470

real time feedthrough, 161

Function Finder Table, 501

function generator, 139

fundamental frequency

pertaining to subharmonics,

198, 199, 200, 283, 284,

471, 478, 486

relating to overtones, 284

fuzz box, 254

gain

control, 237

Gain and Balance, 429

Gain Change, 125

Gain Change feature, 125

gain control, 30

Gain Normalize, 345

Gain Riding, 412

Gamma Scaling

Spectrogram, 312

Gang

Amplitude controls, 237

garbled voice recordings, 221

General Information, 459

Genre

DCTune Library, 81

German, 527

Glossary, 459

Graphic Equalizer, 224

Grayscale, 312

Greyed out items, 66

grit

guitar effect, 253

ground loops, 416

GSM

Cell Phone Noise Filter, 332

GSM Cell Phone Noise, 406

Half Speed Re-Mastering, 428

Hamming, 354

Hams, 57

Hanning, 354

Hardware Connections, 22

Harmonic Distortion, 141

Harmonic Exciter Mode, 255

Harmonic Number, 199

Harmonic Reject, 198

Harmonic Reject filter, 198

Help Menu, 382

Hertz, 197

Heterodyning, 62

hexadecanoic acid, 423

Hi Noise Mode

VVA, 257

High Pass Filter, 212

High Precision Spectrum

Analyzer, 358

Highlight, 197

Direct Spectral Editor, 116

DSE, 116

hill and dale, 234

Hind Quaternion, 177

Hiss, 472

Hiss Control, 160

hissy – sibilant sounds, 284

History, 547

Hot Key Index, 513

How To Do Just About

Anything, 386

Hum Selector, 160

Human Hearing Frequency

Response vs. Age, 515

HVAC, 282

ID3V2 mp3 tags, 97

- Ignition Noise / Static, 62
- IIR**, 472
- IIR Checkbox
 - Harmonic Reject, 200
- Import Playlists, 94
- Impulse Noise filters, 170
- Impulse Noise Generation, 413
- In-band**, 289
- inches per second, 392
- inflection points, 189
- Input Device, 399
- Insert at Start, 109
- Installing
 - the software, 12
- instantaneous vector angle, 366
- Intelligibility of
 - Forensics recordings, 236
- intermodulation distortion, 190
- interpolation, 109
- Inverse Palette**
 - Spectrogram, 312
- Invert, 252
- IPS**, 474
- Jitter Correction, 99
- jittery**
 - display or cursor, 456
- Jukebox, 293
- Kaiser, 364
- Kaiser-Bessel, 364
- keep residue, 293
- Keyboard accelerators, 513
- KT-66, 260
- L/R buttons, 151
- Landscape
 - printer, 103
- laptops, 25
- Large File Conversion to .wav
 - Expanded File Conversion, 69
- laser diode, 78
 - data disc burner, 78
- latency, 163
- lateral cut, 233
- leaky, 293
- License Agreement, 559
- LIMITED WARRANTY, 560
- limiter, 264
- line frequency, 404
- Line Level, 23
- Lissajous**, 475
- Lissajous figures, 368
- Live Mode, 57
- Live Preview, 163
 - Multifilter, 161
- load capacitance, 420
- Lobe Width, 353
- Log to Disk, 165
- logarithmic, 435
- Long Gap**
 - Checkbox, DSE, 118
- Loudness Contours, 230
- Low Pass Filter, 205
- LP, 233
- M3U
 - export file format, 91
 - Import File Format, 90
- magnetic phono cartridge, 395
- Make Destination the Source, 101
- Make Waves, 139
- Manual De-Clicking, 106
- Manual De-Clicking with Paste Interpolate*, 109
- Manual Splitting, 441
- Mark Silent Passages, 344
- Marker Reaction Time, 146
- Markers, 339
- masked, 53
- masking, 281
- Matched Filter
 - algorithm, 333

- Maximum Harmonic
 - Harmonic Reject, 199
- Measurement Table, 515
- Median Filter, 218
- Mem (Memory) Button, 364
- mHz, 140
- Microcassette Tape Start-Stop
 - Detector, 405
- microphone
 - connection(s), 25
- microphone pre-amp, 419
- millihertz, 140
- Minimum Duration, 131
- Mix control, 256
- Mixing two audio files, 112
- MME, 147
- Monophonic**, 476
- MOSFET, 261
- MP3 decoder, 69
- Mp3 Encoder, 93
- MP3 Encoder
 - MP3 encoder setup, 149
- muddiness, 212
- muddy bass, 267
- muffled conversations, 298
- muffled forensics recordings
 - Overtone Synthesizer, 284
- Mu-law (μ -law) Compression**
 - Mu-Law Compression, 476
- MultiFilter, 161
- Multi-path distortion, 267
- multiple notch, 198
- Multiplex Noise, 62
- Musical Scale, 519
- Mute, 120
- Mylar and Polyester backed
 - Tape, 391
- NAB
 - LP EQ Curve, 245
- Narrow band mode**, 337
- Narrow Crackle Filter, 181
- negative feedback, 259
- Noise Gate, 263
- Noise Print, 196
- noise threshold, 196
- Noisegraph®, 160
- noiseprint, 186
- non-linearity, 254
- Normalized Least Mean Squared, 295
- Notch Filter, 215
- NUDGE
 - Hotkeys, 514
- Nudge Size, 145
- Ogg Vorbis, 67
- Ogg Vorbis tags, 445
- Open Destination, 100
- Open Source, 67
- Operating Modes, 27
- Operating point, 255
- Operating Point**, 254
- Optimal Power Calibration**
 - CD Burner Laser, 348
- OSHA Noise Standards, 358
- out-of-band noise, 209
- output level, 254
- Output Mix, 252
- overdriven, 126
- Overlap**, 192
- Overlap %, 194
- overload indicator, 225
- Overtone Synthesizer, 284
- Paintbrush
 - Icon for DSE, 115
- Paper Backed Tape**, 390
- Paragraphic Equalizer, 228
- parametric equalizer, 398
- Paste, 108
- Paste As A New File, 114
- Paste Bleep

- (440 Hz Tone), 114
- Paste Crossfade, 112
- Paste Insert, 112
- Paste Interpolate, 109
- Paste Over, 111
- Paste Silence, 114
- Patches, 492
- Pathe, 234
- Pattern Matching
 - algorithm, 333
- Pausing and Resuming Playback, 136
- Peak files, 381
- Peak Hold**, 370
- Pencil, 115
- Pencil Tool*, 115
- Pentode**, 259
- Phase Inversion, 237
- phase inverter, 260
- phase jitter, 366
- phasing, 418
- ping-pong, 267
- Pink Noise, 139
- Pitch Shift**, 278
- Pitch Shift Mode**, 279
- plate, 468
- Play, 135
- play CDs, 93
- Play Edit**
 - DSE, 118
- Play Looped, 137
- Playback Controls, 128
- Playlist, 89
- Playlists, 87
- PLS
 - export file format, 91
- Point, Click and Measure, 365
- Polynomial Filter, 297
- Pops, 170
- Portrait
 - printer, 103
- Power Amplifier**, 479
- preamplifier, 255
- Pre-Emphasis
 - Voice ID, 320
- Preferences, 144
- Preserve Original**, 343
- Preset Listings, 533
- Preset Manager, 152
- Presets, 152
- Presets Sharing Forum, 492
- Preview Buffers, 450
 - Soundcard Preference, 147
- Preview Buffers (2 to 50), 147
- Preview function, 212
- Preview Mode**, 214
- Print, 102
- Print Preview, 103
- Print Setup, 103
- Printing A Screenshot, 103
- Product Model Number
 - Nomenclature, 558
- product updates, 12
- progress bar, 145
- propagation delay, 415
- Punch and Crunch, 270
- Pure Tones**, 480
- push-pull, 259
- quantization errors
 - minimization thereof, 162
- Quantize, 342
- Quantize for CD Audio, 342
- Random Impulse
 - Random Impulse Generation, 413
- random Noise, 354
- random play, 86
- Range (dB)**, 305
- rasterization, 102
- RC coupled, 258

RCA, 494
 real time, 18
Real Time Analyzer, 481
 real time feedthrough
 Live mode - Multifilter, 161
 Rebuild Peak File, 381
 record speed, 482
 Recorded telephone
 conversations, 437
 Recording Audio, 130
Recording Device, 131
Recording Level Meters, 130
 Recording Resolution, 132
 Rectangular
 Window, 353
 Rectangular Windowing, 354
 Red Book Audio CD's, 94
 Reference Signal, 294
 registration code, 12
 Registration Code
 (finding it in software), 385
 Re-Initialize on Play
 Soundcard Preference, 147
Release Time, 188
 Removing a lead vocal, 420
 Re-Number Markers, 341
 repeated playing
 DCTune Library, 88
repetitive noise, 289
Replace
 Direct Spectral Editor, 117
 DSE, 117
Replacement Bias
 DSE, 117
 Reporting a Problem, 458
 Resistor Color Code, 520
Resolution, 481
Resolution Chart, 351
 resolution conversion, 142
 resonance, 418
 Restoring an old 78 rpm
 recording, 421
 Restoring the Demo, 383
 Reverb, 248
 Reverse File, 266
 Reverse RIAA, 419
 Reverse RIAA Mode, 244
 Rewind, 129
RIAA, 482
RIAA Vinyl LP / 45, 243
Right Mouse Button, 482
 Rip CD Tracks, 94
 RMA, 520
 Roll-off, 417
 Rotary Head, 518
 RPM, 521
Rumble Filter, 241
 rumble reduction, 438
 S/PDIF, 26
 sample
 .wav files, 389
Sample Noise, 45
 Sample Rate, 131
 Sample Spectrum, 301
 Save, 74
 Save Edit Session, 145
 Save Source As, 74
 Scratch, 171
 scratches, 170
 Screen Saver, 27
 Scroll Waveform, 146
 Scrub Audio, 129
 scrubber, 129
 search engine
 DC Tune Database, 85
 Search for help, 385
 Security Tips
 Forensics Audio Labs, 407
 Seismic Noise, 139
 Select All, 114

- selective filtering, 280
- Serial Number
 - (finding it in software), 385
- servo motor, 280
- Shortcuts
 - to Demo Wavefiles, 387
- Shuffle Play, 86
- sibilance, 466
- signal generator, 139
- Silent Passages, 343
- Simulate stereo, 440
- Sine Wave, 141
- Sine Waves, 139
- single file operations, 104
- single-ended, 201
- Skins and Themes, 377
- slider controls, 202
- Slope, 207
- slope control, 323
- slot filter, 216
- Slot Filter**, 484
- slow left-mouse double-click, 96
- Smooth Spectrum
 - spectrum analyzer, 361
- Smoothing Checkbox, 194
- Smoothing Mode, 191, 192
- Snap Selection to Zero Crossing, 124
- Solo/Brass**, 172
- Sound Card, 147
- Sound Card Selection, 21
- Sound Level**, 484
- sound-print, 306
- Source and Destination mode, 28
- Source window, 42
- span, 138
- Spanish, 527
- Special effects, 209
- Spectral Difference, 301
- Spectral Copy, 301
- Spectral Editing, 115
- Spectral Enhancer, 435
- Spectral Filter, 300
- spectral inverse filter, 300
- Spectral Matching, 301
- Spectral Subtraction Mode, 185
- Spectrogram, 310
- spectrographs, 309
- Spectrum Analyzer, 349
- speech clarifier, 300
- Speech Filter, 437
- Speed Change, 273
- Splash Screen, 148
- Spline, 299
- Split and Recombine, 151
- Square Wave**, 486
- Square Waves, 139
- stability, 295
- Standard Precision Spectrum
 - Analyzer, 349
- Stanton 500 RIAA
 - Compensation Curve Preset for the Multifilter, 415
- Start Frequency, 363
- Starting Pitch**, 274
- Static, 62
- Status Bar, 380
- Step By Step Guide, 38
- Stereo Reverse**, 232, 454
- Stereo Simulation, 236
- stereo system, 451
- sticky shed, 391
- Stop Band, 353
- Stop Frequency, 363
- Strength control, 324
- Stretch and Squish, 277
- strobe disc, 521
- Stroboscope**, 486
- Stroboscope Disc Metafiles
 - Printable Strobe Discs, 521

- stuttering, 42
- styli list
 - (styluses), 427
- Stylus**, 426
- Subharmonic Synthesizer, 283
- Support, 448
- SW Radio**, 58
- Sweep, 139
- Sweep Type, 280
- SWL, 451
- Sync Files, 376
- System Requirements, 19
- tag information, 90
- Tag Support
 - Various File Types, 445
- TAO, 347
- Tape Authentication, 403
- Tape Dropout Repair, 393
- Tape Hiss, 392
- target signal**, 289
- TDAF, 292
- Tech Support, 448
- Telephone Jack Wiring, 495
- temp files
 - Fast Edit, 28
- tempo, 277
- Temporary Wave files, 144
- terminology, 527
- THD meter, 355
- Themes
 - Skins and Themes, 377
- Threshold, 400
- thuds, 170
- thump, 183
- Ticks, 170
- Tile, 382
- timbre, 218
- Time Axis, 374
- Time Bracketed Play, 137
- Time Compression, 277
- time constant, 188
- time delay, 293
- Time Display, 369
- Time Domain Adaptive Filter, 292
- time domain interpolation, 109
- Time Format, 148
- Time Offset, 235
- Time Offset slider, 331
- Time/date Stamp, 165
- Time/Date Stamp, 165
- Timer Recording, 137
- Tip of the Day, 383
- Title
 - DCTune Library, 81
- tone arm, 233
- Toolbars, 378
- Touch Tone, 402
- Tracer Technologies**, 489
- Track At Once**, 347
- Tracking, 175
- Tracking Problems**, 428
- Transfer Function, 298
- transformer coupled, 261
- transient response, 192
- Triangle Waves, 139
- Triangular
 - Windowing, 364
- Triangular High Pass, 143
- Trouble Shooting, 448
- TRS
 - Balanced Audio Circuits, 494
- tube, 253
- Tube Types, 258
- turnover frequency, 417
- Turnover Frequency Chart, 523
- turntable, 417
 - connections, 24
- Tutorials, 387

- Ultra High resolution Forensics mode, 355
- Undo, 104
- Undo Procedure Using Fast-Edit Mode, 105
- Universal Mode**, 177
- Universal Serial Bus, 495
- Upgrading Information, 12
- USB, 495
 - Ripping to, 97
- Vacuum Tube Simulator, 253
- Variable Bit Rate (VBR), 68
- variable bit rates, 69
- Variable Frequency Mode
 - Punch & Crunch, 273
- VBR, 149
- vector angle, 366
- Vertical Cut**, 489
- Video/Audio Extraction, 71
 - Video Extraction, 71
- View Menu, 348
- Vinyl LP Mode, 176
- Virtual Preamplifier, 238
- Virtual Valve Amplifier, 253
- Vista**
 - file paths, 457
- voice disguiser
 - forensics Voice Garbler, 335
- Voice Garbler
 - forensics, 335
- Voice ID
 - forensics, 317
- Voiceprint Demo Files**, 389
- Volume Control**, 240, 371
- Vorbis, 67
- vowel
 - formant, 318
- VOX Recording**, 131, 133
- VPA, 238
- VU Meter, 370
- VVA
 - Virtual Valve Amplifier, 254
- WDM, 147
- Weighting Function**, 220
- Welsh, 364
- Wet, 252
- Which Tool Do I Use?, 61
- White to Pink Noise Converter, 142
- Width (or Q):
 - Harmonic Reject, 199
- Window Menu, 381
- Window Selection, 353
- Windows 7
 - file paths, 457
- Windows XP**
 - file paths, 457
- Wire Table, 524
- WLS
 - import file format, 90
- WMA, 67
- WMA File Encoding, 91
- Workspace, 66
- Worldwide Dial Tone
 - Frequencies, 499
- Write CD Text**, 347
- WYSIWYG
 - Printing, 102
- XLR**
 - Balanced Audio Circuit, 494
- XY Display, 365
- Z Axis, 312
- Zero Crossing, 124
- Zero-Pad techniques, 315
- Zoom In, 373
- Zoom Out, 375

Notes

Notes



Printed in the U.S.A.